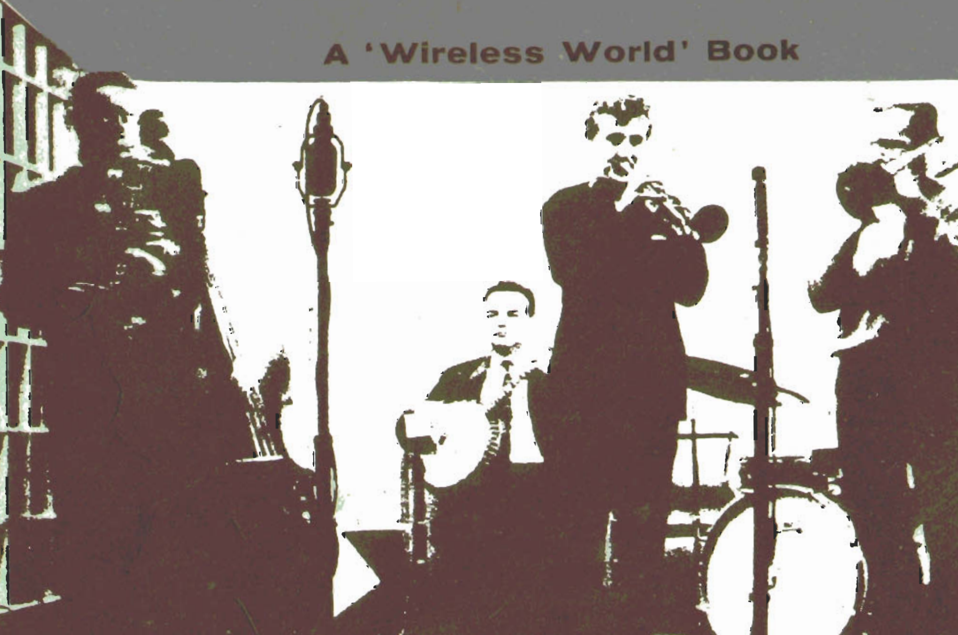
HIGH-QUALITY SOUND PRODUCTION AND REPRODUCTION

BBC Programme Operations Training Manual

H Burrell Hadden

A 'Wireless World' Book



About this book

This book was produced in the Central Programme Operations Department of the BBC for their own personnel, both technical and non-technical, to enable them to obtain the best results from studio equipment. But it will also prove of absorbing interest to other broadcasting organisations and people, both amateur and professional, interested in the production and reproduction of high-quality sound.

The book is divided into three parts, the first of which deals with the basic principles of sound and electricity and includes chapters on the theory of musical instruments and studio acoustics.

The second part describes studio equipment. Microphones, loudspeakers, studio control desks, outside broadcasting equipment and P.A. equipment are among the topics fully discussed.

The final section deals with the important subject of the placing of microphones for talks, vocalists and all types of bands, orchestras and musical groups. Other subjects covered are the control of volume, the production of sound effects and stereophony.

The book is lavishly illustrated throughout with half-tones and line drawings—the latter bringing a novel approach to the pictorial presentation of complicated information in a most simple form.

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1

BASIC SOUND THEORY

1.1. NATURE OF SOUND WAVES

The first three chapters are given up to a brief summary of those aspects of acoustic theory which are most useful in the business of collecting sounds in the studio and reproducing them in the listener's home. We begin by building up a picture of sound

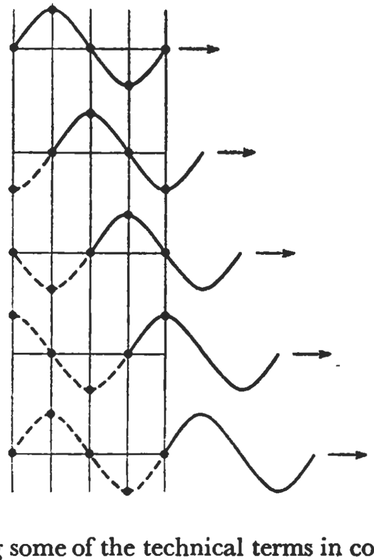


Fig. 1.1. Horizontal movement of a water wave resulting from the vertical vibration of individual particles

waves in action and defining some of the technical terms in common use. Consider first the mechanism of waves on the surface of water.

Drop a stone into a still pond (the analogy in terms of sound would be a hand-clap or an explosion) and a wave will be seen to travel outwards in an ever-increasing circle, the wave taking the form of a disturbance on the water's surface. Drop a succession of stones, or plunge something up and down in the water (analogous with a

continuous sound), and a continuous rippling outwards will take place, capable of setting a line of small corks into *up-and-down* vibration.

In Fig. 1.1, the wave is represented in successive drawings as moving to the right, while the corks—which we have introduced simply as markers or guides to the water vibration—move up and down on vertical lines just as if they were suspended from a spring. The natural time delay for greater distances from the source means that successive corks reach the top of their swing a little later, so that the crests of the wave appear to travel outwards. In fact, there is no outward movement of the corks and water at all.

Set a tuning-fork in vibration and waves of energy will travel outwards. This is a *sound wave*. It takes the form of a disturbance in the air (or other medium through which it travels), and is capable of setting a thin membrane, such as the ear-drum, in *to-and-fro* vibration. As with the water waves, the air particles do not travel out with the wave. They imitate the fork vibrations, oscillating " on the spot ", and pass on the energy of the wave by

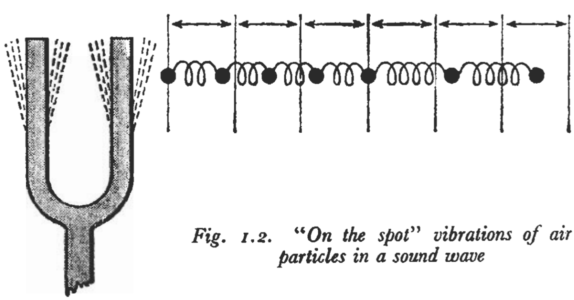


Fig. 1.2. "On the spot" vibrations of air particles in a sound wave

reason of the elastic coupling that exists between them (shown in Fig. 1.2 as coupling springs).

The corks which we used as guides to the water vibration are not going to be of much use to us here, but the history of positions taken up by the tuning-fork may be recorded by fastening a tiny pen to it, and moving paper underneath. The wavy trace, by introducing the appropriate delay in time, might be a record of the movements of any of the air particles in the path of the wave.

Fig. 1.3 will repay closer examination. Starting at the middle point, A, we see that the fork moved out to the right, then swung

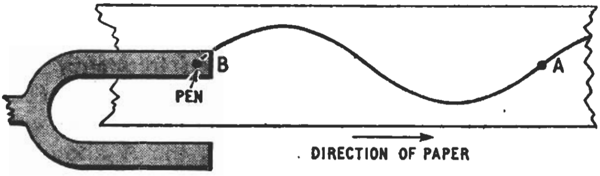


Fig. 1.3. Waveform produced by tuning-fork on moving paper

back through the midpoint and fully out to the left. It then returned to the midpoint, at B, and began to repeat this sequence of movements all over again. (The result is a sine wave, showing that the fork is performing simple harmonic motion, as described in the Appendix.)

1.2. FREQUENCY AND PITCH

1.2.1. Frequency

The frequency of the fork vibrations is defined as the number of complete vibrations (or cycles) per second. Thus, if 440 complete vibrations are performed in one second, the frequency would be written 440 c/s. Where very high frequencies are being considered we use kilocycles per second and megacycles per second as units. Thus, 5,000,000 c/s may be written as 5,000 kc/s or 5 Mc/s.

In point of fact, although vibrations at any frequency will give rise to sound waves, the human ear is only sensitive to sounds in the frequency range 20 c/s to 20,000 c/s approximately. More will be said about this in a later chapter.

1.2.2. Pitch

The pitch of a musical note may be defined as that property by which we place the note as being high or low on the musical scale.

Fig. 1.4. Standard Musical Pitch A



Pitch and frequency are related, and increasing or decreasing the rate of vibrations will cause the sound to move up or down the musical scale. For example, by agreement in 1939, Standard Musical Pitch A, which is shown in musical notation in Fig. 1.4, has

been fixed at 440 c/s, and any structure vibrating regularly at this rate, violin string, air column, or circular saw, will emit this note.

A discussion of the connection between frequency ratios and musical intervals will be reserved for the next chapter, but the simple rule of doubling the frequency to raise the pitch by an octave will already be known to most readers—e.g., the A above Standard Musical Pitch A is 880 c/s.

1.3. HARMONICS AND MUSICAL QUALITY

The tuning-fork, an invention of John Shore in 1711, is well known for its ability to maintain vibrations purely at one frequency. Most other sound sources, musical instruments included, tend to perform several modes of vibration simultaneously, so that the sounds which result are not pure tones, but complex. Usually the various component frequencies form a family, or series, being simple multiples (twice, thrice, etc.) of the lowest frequency present. This last is

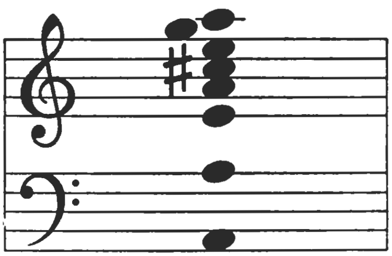


Fig. 1.5. First eight harmonics of 110 c/s shown in musical notation

called the *fundamental frequency*, and it decides the apparent *pitch* of the note. The higher frequencies, or overtones, are called *harmonics*.

Thus, a double bass sounding A two octaves below 440 c/s will contain, mixed in certain proportions, the following family of frequencies:—

fundamental	110 c/s
2nd harmonic	220 c/s
3rd harmonic	330 c/s
4th harmonic	440 c/s
5th harmonic	550 c/s

and so on (Fig. 1.5).

It is the presence of harmonics in varying numbers and relative strengths which give the sounds from different musical instruments

their characteristic quality or *timbre*. We recognise the clarinet, for example, partly because of the preponderance of odd harmonics (third, fifth, etc.) which that instrument produces.

The *waveform* corresponding to the pure vibration of a tuning-fork was seen in Fig. 1.3 to be a simple sine wave. When several frequencies are present together the resultant waveform, if displayed on

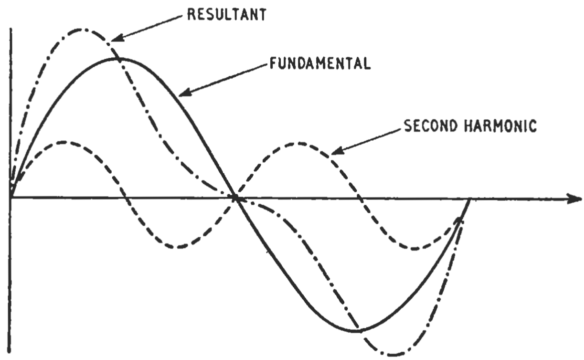


Fig. 1.6. Waveform produced by combining fundamental and 2nd harmonic

a cathode-ray tube, for example, is found to be more complex. Generally speaking, the peakier the waveform, the higher the harmonic content. A saw-toothed waveform may contain the complete harmonic series.

Fig. 1.6 shows in diagrammatic form the derivation of the waveform produced by a fundamental and second harmonic which are in phase (for definition see section 1.5) and whose amplitudes are in the ratio of 3 : 1.

1.4. COMBINATION TONES

When a number of frequencies are sounding simultaneously, the brain receives the impression that additional frequencies are present. This effect is inherent in the mechanical action of the ear, and makes analysis of complex sounds in terms of the sounds actually imagined in the brain exceedingly difficult. One of the "extra" sounds created in this way, when two pure tones are being sounded together, is at a frequency which is the difference between the two real frequencies. For example, if tones at 1,000 c/s and 900 c/s are present, the hearer feels that 100 c/s (1,000—900) is also sounding.

This particular combination tone is called the *difference frequency*. When two real frequencies differ by only a few c/s, the listener receives the impression of pulsations at the difference frequency, which is usually called in this case the *beat frequency*. The process of tuning two notes to exact unison is thus one of eliminating beats.

1.5. PHASE

Phase is the term used to describe the stage reached by a vibrating particle in its cycle of movement. Phase is usually measured in degrees, 360° corresponding to one complete cycle (see Appendix).

1.6. WAVELENGTH

The distance, measured along the wave, between successive particles which are *in phase*, is called the *wavelength* (symbol λ)—e.g. the distance from crest to crest. Now the wave will travel this distance in the time it takes the source to perform one cycle, and it will travel f times this distance in one second, if there are f cycles

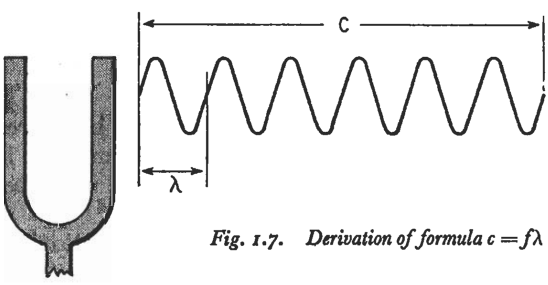


Fig. 1.7. Derivation of formula $c = f\lambda$

per second. Therefore, if the frequency of the tuning-fork in Fig. 1.7 is f c/s, and the wavelength is λ feet, the distance travelled by sound per second is $f \times \lambda$ feet per second.

The distance travelled per second is called the *velocity* (symbol c), and so we have the important formula

$$c = f\lambda$$

which is true for all types of waves (Fig. 1.7). The velocity of sound in air is approximately 1,120 ft/sec, and the formula becomes $1{,}120 = f\lambda$, from which it is possible to calculate the wavelength in air for any frequency, and vice versa.

Example 1

$$f = 112 \text{ c/s}$$

$$1{,}120 = 112\lambda$$

$$\therefore \quad \lambda = \frac{1{,}120}{112} = 10 \text{ ft}$$

Example 2

$$f = 1{,}120 \text{ c/s}$$

$$1{,}120 = 1{,}120\lambda$$

$$\therefore \quad \lambda = \frac{1{,}120}{1{,}120} = 1 \text{ ft}$$

Example 3

$$f = 11{,}200 \text{ c/s}$$

$$1{,}120 = 11{,}200\lambda$$

$$\therefore \quad \lambda = \frac{1{,}100}{11{,}200} = 1/10 \text{ ft}$$

Example 4

What is the range of wavelengths in air corresponding to the range of normal hearing?

Lowest frequency is 20 c/s

$$\therefore \quad \lambda = \frac{1{,}120}{20} = 56 \text{ ft}$$

Highest frequency is 20,000 c/s

$$\therefore \quad \lambda = \frac{1{,}120}{20{,}000} = 0{\cdot}056 \text{ ft}$$

$$= 0{\cdot}67 \text{ in.}$$

1.7. FACTORS AFFECTING SPEED OF SOUND

1.7.1. Temperature

The velocity of sound waves is virtually independent of frequency. If this were not so, the various notes from an orchestra would reach the audience in a hopeless muddle. It is found, however, that temperature has a direct bearing on sound velocity, and c increases by about 2 ft/sec for each degree Centigrade rise in temperature.

The velocity at 0° Centigrade is 1,087 ft/sec, and the velocity at any temperature T can therefore be calculated from the formula

$$c = 1{,}087 + 2T \text{ ft/sec}$$

An example of this effect occurs in the tuning of wind instruments. The wavelength of the notes from a wind instrument is decided by the length of the air column. If the velocity increases due to warming of the air, this will tend to sharpen the pitch of the notes. It is important, therefore, when tuning, that such instruments be brought up to the temperature in which they are to be played. A temperature change of 15° Centigrade results in a shift of about one semitone.

1.7.2. Humidity

The humidity, or moisture content of the air, has only a small effect on sound waves. The velocity is about 3 ft/sec faster in saturated air than in dry air.

1.8. INTENSITY AND LOUDNESS

So far we have restricted our discussion to the rate of vibration, or frequency, of the source, and the nature and speed of the waves. It is now necessary to examine the behaviour of the source when the *strength* of the vibrations is varied. One obvious measure of the strength of the vibration is the distance which particles of the source swing to either side of their average position. This is called the *amplitude*, and more violent vibrations of greater amplitude result from striking, bowing, etc., harder, which corresponds to putting in more energy. This naturally causes a greater amount of sound energy to be radiated. The amount of energy radiated per second is called the *power* of the source.

The *intensity* of a sound due to a given source is proportional to the power of the source and inversely proportional to the area over which the sound is spread.

The *loudness* * of a sound is dependent on the power of the source, and on the nature of the particular sound. By " nature " is meant the total composition of the sound at the position in which

* Loudness as defined above is a quantity which can be measured. The subjective aural loudness actually heard will include these quantities, but will also depend on a number of other factors. This " apparent loudness " will change with the level of background noise; the particular person hearing it; and the psychological and physiological condition of the person at the time. These factors, outside the control of a broadcasting organisation, make the choice of transmission levels, for example between speech and music, very difficult (see chapter 14).

it is heard; namely, its frequency content, and the effect of the acoustic environment.

1.9. FREE VIBRATIONS AND DAMPING

When a sound source such as a tuning-fork or violin-string is struck, and left to itself, it performs what are called *free vibrations*. These will take place at a frequency—the *natural frequency*—which is determined by the mass and springiness of the source and by nothing else. In the same way, a pendulum will swing to and fro at a rate (frequency) which is a function of the length and of nothing else.

The rate at which the vibrations die out depends on the rate of expending energy (defined above as the power), which is made up of the radiated energy plus the energy lost in overcoming air and other friction—called the *damping*.

Thus, a tuning-fork held in the hand decays very slowly, and is said to be lightly damped. When placed on a table or box, the fork radiates more energy per second, and comes to rest more rapidly. Similarly, a garden swing will soon come to rest if we scrape our feet on the ground, and the dampers on a piano silence the notes almost immediately.

1.10. RESONANCE

1.10.1. Forced Vibrations and Resonance

If an alternating force of, say, 500 c/s is applied to a tuning-fork whose natural frequency is 400 c/s, the vibrations which result will be at 500 c/s and will be of very small amplitude. These are called *forced vibrations*.

If the driving frequency is then gradually lowered, the fork vibrations will follow it, since all the energy is being supplied by the driver, and the amplitude of the fork vibrations will be found to increase. When the driving frequency coincides with the fork's natural frequency (400 c/s), very great amplitudes will result. Further reducing the driving frequency causes the fork amplitude to fall again to a very small value.

The increased response of a vibrating system, whether it be mechanical, acoustical or electrical, system when driven at its own natural frequency is called *resonance*, and so the natural frequency of a system is sometimes called its *resonant frequency*.

1.10.2. Examples of Resonance

Some examples of resonance are:—

(1) shattering a wine glass by singing or playing its "ringing" frequency;

(2) starting heavy bells by pulling lightly on the rope once in each cycle;

(3) unwanted resonance in the air contained in studios at frequencies which are determined by the dimensions;

(4) the boominess of some loudspeaker cabinets.

1.10.3. Sharpness of Resonance

If we plot a graph of amplitude against frequency, a peak is found at resonance with falling response on either side. The sharpness of this resonant peak depends on the amount of damping—or friction—present. A lightly damped system is said to be

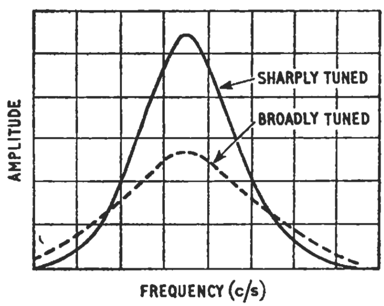


Fig. 1.8. Sharp and broad resonance

sharply tuned, and will respond only to frequencies near resonance—e.g. a tuning-fork.

A heavily damped system is *broadly tuned*, and will respond over a wide band of frequencies—e.g. a piano sounding-board (Fig. 1.8).

Occasionally it is necessary to design a sharply tuned or highly selective system, such as the resonator tubes below the bars of a xylophone. But a more common requirement is a flat response, with either the amplitude or the velocity of the system maintained substantially constant over a range of frequencies. For example, in the design of loudspeakers, large diaphragm resonances leading to the exaggeration of certain frequencies are as far as possible avoided. In fact it is not possible to make a diaphragm completely free from all resonances and the response curve usually shows a series of tiny peaks. The violin, on the other hand, uses

the resonances present in the body of the instrument to produce its characteristic sound.

Throughout the above it has been assumed that the driving system was more massive or powerful than the driven system. In cases where the two components in a *coupled system* are of comparable mass, the resultant frequency may be intermediate between the resonant frequencies of the two, approximating more closely to that of the heavier system—e.g. the reed and the air column of a wood-wind musical instrument compete for control of the pitch of the notes, but only in the case of the saxophone (heavy reed) can the reed take control easily.

1.11. DIRECTIONAL PROPERTIES OF SOUND SOURCES AND THEIR EFFECT AT A DISTANCE

Power was defined earlier as the rate of sending out energy, and a great range of measurements is found to exist. For example, the maximum power radiated by a large orchestra is about 70 watts, while for ordinary conversation it is 0·0001 watts. Whether sounds carry over long distances from a given source, or quickly become too faint to hear does not solely depend on this initial power content. It is largely a question of *directivity*. If sounds are to be radiated efficiently in a given direction, they must be beamed or focused in that direction at the expense of other directions. (Consider the old-fashioned speaking-tube.) Radiating equally in all directions results in dissipation of the available energy, and a rapid falling off in volume.

Directivity, in turn, is very much bound up with the size of the sound source in relation to the wavelength of the sound radiated. To clear up this important point, let us take the case of (a) a very small, and (b) a very large source (Fig. 1.9):—

(a) *Very Small Source:* in this case, the wavefront takes the form of a continually expanding sphere—hence the name *spherical waves*—and the power radiated is spread over a wider area as the distance increases. Now the intensity is calculated by dividing the total power by the area of wavefront, and so it is continually diminishing with increasing distance. The surface area of a sphere increases in proportion to the radius squared, and so we have the intensity falling off in this ratio, i.e. *intensity falls off with distance squared.*

(b) *Very Large Source:* one way of treating this case is to regard each of the particles of the large source as a point source, when

the effective wavefront of the combination (allowing for expansion at the fringe) is seen to be a flat area which proceeds outwards at right angles to the face of the source—hence the name *plane waves*. Thus, from a very large source *intensity falls off very little with distance.*

Waves from point sources tend to become plane at great distances from the source. Conversely, waves from plane sources (e.g. line

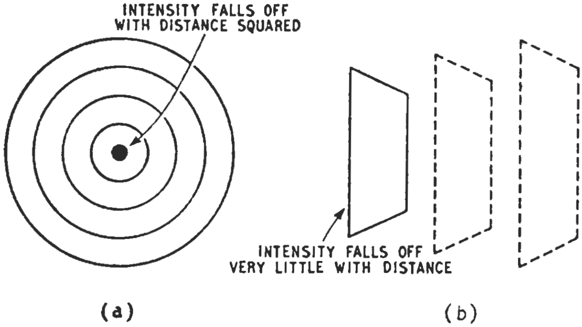


Fig. 1.9. Diagrammatic representation of sound radiation from (a) very small, (b) very large sources

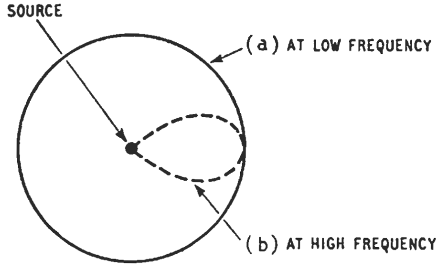


Fig. 1.10. Polar diagram of radiation from a typical source

source loudspeaker, Chapter 9) tend to become spherical at great distances.

In the above theoretical treatment, one frequency only was considered. Musical instruments, loudspeakers, etc. are, however, required to radiate a wide range of frequencies as evenly as possible. The criterion as to which of the above modes operates is found to

be the relation between the wavelength of the sounds and the dimensions of the instrument.

When the length and breadth of the instrument are smaller than the wavelength, it emits approximately spherical waves with consequent low efficiency radiation in any given direction. At high frequencies, however, where the wavelength is small, the instrument becomes progressively more directional, and the " carrying power " is increased, since all the energy is concentrated in one direction instead of being dispersed over a wide area. An example is given in Fig. 1.10 where the curves show the distribution of sound pressure around a circular source (a) at low frequency when the diameter of the source is smaller than the wavelength, and (b) at high frequency when the diameter is longer than the wavelength. All points on curve (a) will have equal sound pressure, and similarly with curve (b). As an analogy one might say they are similar to the contour lines of a map, which show points of equal height above sea level.

1.12. SUPERPOSITION OF SOUND WAVES

When a number of sound waves travel through the same region of air, an interference pattern is set up in the common region, each wave continuing on without losing its identity. The resultant positions taken up by the air particles in the path of both waves are calculated by adding the displacements due to each of the waves considered separately.

For example, it is fairly well known (and easily verifiable) that four regions of comparative silence exist along lines at 45° from a tuning-fork. The prongs of the tuning-fork form a double source in anti-phase with cancellations taking place along the directions indicated.

A more important example of the combination of sound waves is the total reflection of a sound back along its original path. The relative phase of the reflected wave is found to be such as to cancel the advancing wave at multiples of half-a-wavelength from the reflecting surface (these points are all called *nodes*), while midway between the nodes the two waves arrive in phase, and reinforcement takes place. (These points are called *antinodes*, or *loops*.)

The interference pattern described above is stationary, and is called a *standing wave*. It may be noticed in passing that the amplitude of a standing wave is twice that of the advancing wave alone, being the sum of the two waves in question. Of course, this

assumes perfect—i.e. 100%—reflection. In general, sounds are not totally reflected from walls etc., as some of the energy is absorbed by, or transmitted through, the wall. Thus, the amplitude of the reflected wave is usually less than that of the advancing wave. When a standing wave pattern is set up in a studio, small adjustments in the microphone position may cause serious changes in the quality of sound.

1.13. OBSTACLE EFFECT

If the dimensions of an object in the path of sound waves are made smaller than the wavelength of the sound, the amount of energy reflected is very small. Bending action appears to take

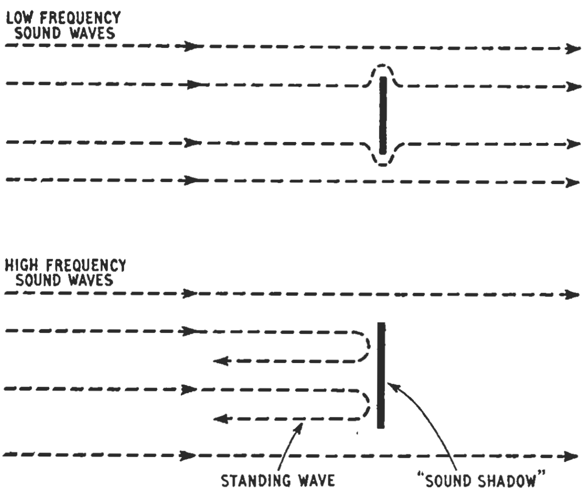


Fig. 1.11. The obstacle effect

place, and the wave tends to carry on beyond the object as if nothing had been in the way. Reducing the wavelength of the sound (increasing the frequency) below the size of the obstacle causes reflection as usual, and if the wave meets the object at right angles, a standing wave will result (Fig. 1.11). This frequency-discriminating property of objects to sounds of different wavelengths is called the *obstacle effect.* Remembering that the range of wavelengths of sound in air extends from about 50 feet down to about 1 inch, we see that everyday objects such as tables, chairs, people, and

studio screens, will tend to reflect some frequencies, but not others. This has an important bearing on the placing of artistes and furniture in the studio if " frequency-discriminated " sound is not to reach the microphone—i.e. if distortion is to be avoided.

Of the many examples that might be chosen, it will be seen that acoustic studio screens call for careful use since their width is about 3 ft. This means that, whereas high frequency sounds are effectively controlled by the screen, sounds of wavelengths greater than 3 ft. (frequencies below 350 c/s) will " bend round " the screen and not be reflected. More will be said about this important subject later.

The behaviour of curved reflecting surfaces is also conditioned by the " obstacle effect ". Reflections from a concave surface, for example, tend to be focused together, but only if the diameter of the reflector is greater than the wavelength of the incident wave. It follows that the parabolic microphone reflectors used sometimes in outdoor broadcasts and recordings—where they are trained on distant objects such as the batsmen in cricket broadcasts, song-birds, etc.—are comparatively ineffective at low frequencies. The two models available are 3 ft. and 18 in. in diameter, and have " cut-off" frequencies of about 300 c/s and 600 c/s respectively.

Similarly, convex surfaces, which are used in studios to disperse or " diffuse " sound waves, are only effective when the dimensions of the surface as a whole are greater than the wavelength.

2

MUSICAL ACOUSTICS

2.1. ANATOMY OF THE HUMAN EAR

Before describing some of the characteristics and limits of hearing, a brief word on the construction of the ear may be useful. The diagram (Fig. 2.1) is purely functional, and not drawn to scale.

The *auditory canal* leads from the opening in the outer ear, is about 1 inch long by ¼ inch wide, and is broadly resonant in the range 2,000 to 6,000 c/s. It is closed at the inner end by the *ear-drum*, an oval of skin stretched over a bone frame. Attached to the inner

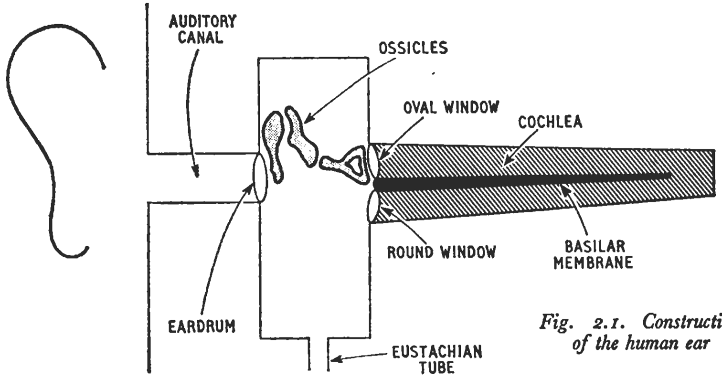


Fig. 2.1. Construction of the human ear

side of the ear-drum is the first of three *ossicles* or tiny bones. These, together with their supporting muscles, etc., are contained in the *middle ear* cavity which is air-filled, and connected to the throat by means of the *eustachian tube*. This is normally closed, but opens on swallowing or yawning, to permit air to enter or leave, thus equalising the long term pressures on the eardrum.

The ossicles provide some protection against sudden bursts of sound, and pass on the vibrations of the eardrum to another thin

membrane, the *oval window*. In this way, the vibrations are conveyed to the fluid in the *cochlea*, the movement being absorbed by the *round window*.

The casing of the cochlea is a bony structure wound in spiral form (shown straightened out in the diagram). For the present purpose, it can be taken to be divided into two galleries by a partition which runs nearly to the apex and which includes the *basilar membrane*. Reaching out into the cochlear fluid, and responding to its movements, are a large number of hair cells extending all along one side of the basilar membrane. These are the endings of nerve fibres running to the brain and which keep the brain informed of the vibratory stimulus to the ear-drum.

The brain uses the information sent to it by the ears in two different ways. Firstly, it enables us to hear the sound; that is, to determine its pitch, quality and loudness. Secondly, it performs a calculating operation on the minute differences between the sounds received at each ear. The result of this is that we are able to determine the position of these sounds in space. This operation will be more fully discussed in the chapter on Stereophony (chapter 16).

2.2. THRESHOLDS OF HEARING

The human ear will respond to a very wide range of *frequencies*, from about 20 to 20,000 c/s—or nearly ten octaves. Sensitivity to high frequencies is found to deteriorate with age.

The minimum *intensity* level that can be heard by a given observer is not the same at all frequencies. In most people maximum sensitivity usually occurs round about 3,000 c/s, with a falling off gradually at lower frequencies, and more steeply at higher frequencies. The graph of minimum audible intensity against frequency is called the *threshold of hearing* (Fig. 2.2).

As an example of the degree of variation in our hearing ability, the power required to produce an audible sound at 50 c/s is about 1,000,000 times that necessary at 3,000 c/s.

If the intensity of a given sound is progressively increased, causing it to get louder and louder, a point is reached where the sensation is one of feeling or pain rather than hearing. This level does not depend very much on the frequency of the sound and is called the *threshold of feeling*.

2.3. THE MUSICAL SCALE

As already mentioned in Chapter 1, any two musical notes which are an octave apart have frequencies in the ratio of 2 : 1. If we

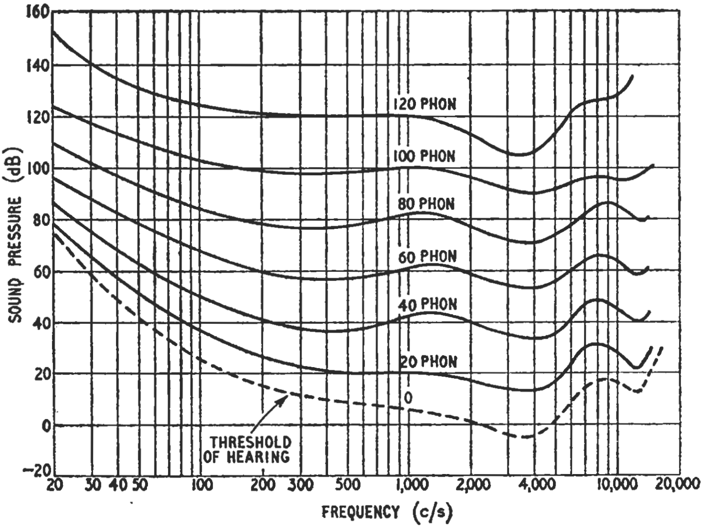


Fig. 2.2. Equal sensitivity curves for average hearing (Robinson and Dadson, N.P.L.)

write down the frequencies for the octaves of A, we have the following result:

A_{iv}	A_{iii}	A_{ii}	A_{i}	A	A^{i}	A^{ii}	A^{iii}
27·5	55	110	220	440	880	1760	3520

The frequencies are in geometric progression with a common ratio of 2. It is for this reason that frequency response graphs etc. are drawn on a logarithmic scale on which equal divisions are allotted to equal changes in the logarithm of the frequency (powers of ten) rather than on a linear scale, where the divisions apply to a fixed number of cycles per second.

The octave is perhaps the most obvious example of a pair of notes which stand in a recognisable relation to each other. But a simple ratio of frequencies is found to exist for all the musical intervals which have a simple relationship to the ear. In fact, the order of *consonance* (or degree of harmony) of two notes sounding together is found to depend on the smallness of the numbers used to express the " frequency ratio ". Next most concordant after the octave are the perfect fifth (3 : 2) and the perfect fourth (4 : 3), and so on.

The interval of a second (9 : 8) is comparatively *dissonant* (or discordant), due partly to beats between the harmonics of the two notes.

Our major and minor scales have evolved from the grouping of notes which provide the largest number of consonant combinations.

For example, the major scale of C, taking the frequency of C as x, may be laid out as follows:—

	C		D		E		F		G		A		B		C
Frequency	x		$\frac{9}{8}x$		$\frac{5}{4}x$		$\frac{4}{3}x$		$\frac{3}{2}x$		$\frac{5}{3}x$		$\frac{15}{8}x$		$2x$
Ratio to preceding note		$\frac{9}{8}$		$\frac{10}{9}$		$\frac{16}{15}$		$\frac{9}{8}$		$\frac{10}{9}$		$\frac{9}{8}$		$\frac{16}{15}$	

This is called the scale of *just intonation.*

The present system to which keyboard instruments are tuned, and most music approximates, is called the scale of *equal temperament.* Here, the octaves are tuned to true pitch, and each octave is divided into twelve equal divisions called *tempered semitones.*

In equal temperament, music can be played equally well in all keys. There is, however, the serious disadvantage that only the octaves are in perfect tune. There is evidence that singers, string-players, etc., when not accompanied by keyboard instruments, tend to depart from equal temperament to something approximating more closely to true, natural intervals.

2.4. THE INTENSITY SCALE

Our appreciation of *loudness* is found, generally speaking, to follow a somewhat different law: if we multiply the *intensity* of sound by 10 the *loudness* is approximately doubled.

If it be increased again by a further factor of 10, that is, 100 times the original intensity, the second change in loudness would appear to the ear to be the same as the first, that is, the sound will now be about four times as loud. The unit which relates these changes in intensity is termed the *Bel* and so 1 Bel is the amount of change when the intensity is changed by a factor of 10.

Now the change in intensity represented by the two thresholds at 1,000 c/s (see Fig. 2.2) is approximately in the ratio of 1,000,000,000,000 : 1—i.e., 10^{12} : 1. This is twelve factors of

ten, and therefore is said to be 12 *Bels*. Thus the number of Bels is the same as the index of the power of 10.

$$\therefore \quad \text{ratio in bels} = \log_{10}\frac{I_2}{I_1}$$

In practice this is too large a unit and the *decibel* is used:—

$$\therefore \quad \text{ratio in decibels (dB)} = 10 \log_{10}\frac{I_2}{I_1}$$

Under good listening conditions the minimum change in intensity that can be detected by the ear as a change of loudness is one decibel.

$$\text{Thus} \quad 1 \text{ dB} = 10 \log\frac{I_2}{I_1}$$

$$0{\cdot}1 = \log\frac{I_2}{I_1}$$

$$\text{antilog } 0{\cdot}1 = \frac{I_2}{I_1} \simeq 1{\cdot}26$$

$$\therefore \quad I_2 = 1{\cdot}26\, I_1$$

$$= I_1(1 + 0{\cdot}26)$$

It will be seen therefore, that a sound of *any* intensity must be changed by 26% before a loudness change can be detected.*

If we refer again to Fig. 2.2, the curves show the intensity in dB required to produce sounds at equal loudness, at varying loudness levels. It will be seen immediately that the intensity varies with frequency. It is evident then that the decibel is not a suitable unit to denote a change in loudness level, since for a given dB change in intensity the loudness is entirely dependent on frequency.

A different unit is therefore used, called the *phon*, and a sound is said to have a loudness level of n phons if it sounds equally loud as a 1,000 c/s tone ndB above the zero level. Zero level is agreed to be the minimum intensity required just to hear a 1,000 c/s note.

* For a discussion of the decibel as a measure of change in an electrical circuit see Appendix A.

Example 1

How many dB separate the thresholds of hearing and feeling at 1,000 c/s?

The ratio of the intensities has already been given as $10^{12} : 1$

$$\therefore \quad \frac{I_1}{I_2} = \frac{10^{12}}{1} = 10^{12}$$

$$\therefore \quad \text{number of dB} = 10 \log 10^{12}$$

$$= 10 \times 12$$

$$= 120 \text{ dB}$$

Example 2

Sounds in a studio are found to die away to one-millionth of their original intensity in one second. What is the decay of sound, expressed in dB?

The ratio of the intensities is 1,000,000 : 1

$$\text{i.e. } \frac{I_1}{I_2} = \frac{1{,}000{,}000}{1} = 10^6$$

$$\therefore \quad \text{fall in dB} = 10 \log 10^6$$

$$= 10 \times 6$$

$$= 60 \text{ dB}$$

Example 3

Given that the logarithm of 80 is approximately 1·9, how many dB more intensity will result from eighty grand pianos playing together than from one?

The ratio of the intensities is 80 : 1

$$\therefore \quad \text{increase in intensity} = 10 \log 80$$

$$= 10 \times 1{\cdot}9$$

$$= 19 \text{ dB}$$

2.5. THEORY OF MUSICAL INSTRUMENTS

All musical instruments have this in common: they are designed to produce one or more pleasing tones. Common to most musical instruments is a system of one or more *resonators*, by which is meant a component possessing a discrete resonant frequency of free vibration. Many structures possess resonant characteristics suitable for use in musical instruments, including strings, air in pipes, bars, membranes and electronic oscillators. In a given instrument, the pitch of the resonator may be fixed, or variable. Some means of exciting the resonators into vibration is provided, under the control of the musician.

Examples of instruments possessing the most common methods of excitation are listed below:

Instruments	*Resonators*	*Excitation*
Violin	Strings (4)	Bow or fingers
Harp	Strings (46)	Fingers
Piano	Strings (88 notes)	Hammers
Flute	Air column	Edge tone
Oboe	Air column	Reed
Trumpet	Air column	Player's lips
Xylophone	Bars	Sticks
Timpani	Drumskin	Sticks

There is another part of many musical instruments that should be mentioned. It is included to improve the radiating efficiency, and may be given the general name—*radiator*. A string, for example, tends to cut through the air as it vibrates without causing much air movement. Hence the need for a sounding board in pianos, and the complicated body of violins, etc.

It follows that the overall quality or timbre of a given instrument is not decided by the harmonic content of the resonant component alone (see Chapter 1), but is likely to be modified by the selective characteristics of the radiator. Many instruments possess a region of pitch—known as the *formant*—in which tones, fundamental or harmonic, tend to " sound out ". Violins, for example, derive much of their characteristic tone from the formant reinforcement over the range 3,000 to 6,000 c/s. The piano is something of an exception in that its very heavily damped sounding board responds

evenly to all frequencies, and radiates the string tone more or less faithfully.

2.6. MUSICAL STRINGS

Strings are used as the tuneable, or resonant, system in many musical instruments.

A string is capable of vibrating in several modes simultaneously, to produce a full range of harmonic frequencies. Fig. 2.3 shows the sine wave form of the standing wave on a string for the fundamental and the first four harmonics. The fixed ends always being nodes, the " harmonic number " is seen to be the number of loops formed along the string.

Only transverse vibrations of the string are considered in music —witness the unmusical sounds produced by the new violinist who

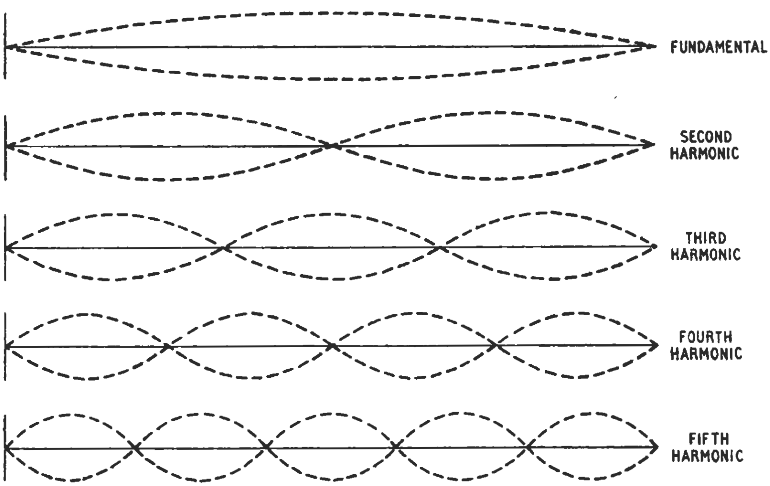


Fig. 2.3. Harmonic modes of vibration of strings

fails to bow at right-angles to the strings, and excites dissonant vibrations along the length of the string.

2.6.1. String Quality

The effect on quality of the formant has already been mentioned, and will vary slightly from one instrument to another. It is not usually under the control of the player. Great variations in the

tone are possible, however, depending on the point and method of attack.

Bowing close to the bridge (*sul ponticello*), for example, is sometimes resorted to for especially brilliant quality, rich in harmonics. A broad bow gives extra mellowness of tone, while using the wooden back of the bow (*col legno*) produces a harsh, dry effect. Again, fairly mellow tone is to be expected when strings are plucked with a round " instrument ", such as the fingers. Using a sharp plectrum pulls the string into the shape of a triangle, and much of the energy is thrown into the highest harmonics, with resultant metallic tone.

As a final example, if a hard, sharp hammer is used, a sharp kink is given to the string, and, in the extreme case, all harmonics are formed at equal strength—e.g. a coin dropped on to piano strings. The felting of piano hammers is calculated to produce a pleasing proportion of harmonics, and " tinniness " begins when the felt becomes matted down.

It is worth noting that the effective hardness of the hammers increases when the notes are played forcibly in forte passages. This means that dynamic variations in piano playing are characterised not only by simple variations in loudness, but by real variations in quality. Louder passages are more brilliant in tone, due to the extra production of upper harmonics.

2.7. MUSICAL AIR COLUMNS

A large group of musical instruments use the resonance of one or more pipes to produce notes on the scale. As with strings, the principles involved and the tuneable factors have been known for very many years. When the air in a cylindrical pipe is excited into vibration, a standing wave is set up due to reflection and re-reflection between the two ends. The length of the pipe is therefore related to the wavelength of the standing wave to which a given pipe will resonate. There is a difference in the relationship, by a factor of 2, depending on whether both ends of the pipe are open to the atmosphere, or only one.

2.7.1. End Correction

Because of the change-over from *plane* waves inside to *spherical* waves radiated from the end, the true position of the antinode is not exactly at the end of a pipe, making the effective length slightly more than the measured length.

The extra length (or end correction) for harmonic frequencies is slightly different from that for the fundamental, a factor which

has some bearing on the tone of different instruments. Wide bore pipes are found to generate fewer harmonics, and sound less brilliant than narrow pipes, due to the mistuning of partials when the end correction is taken into account.

2.7.2. Methods of Excitation of Air Column

The air column in a wind instrument may be set in vibration in at least three ways, namely:—

(a) Edge tones

(b) Reeds

(c) Player's lips

(a) *Edge Tones* are produced when a stream of air is directed on to a solid lip or wedge. If this is coupled to a resonant pipe, excitation takes place at the natural frequencies of the pipe, eddies being struck from alternate sides of the wedge. Typical edge tone instruments are the flue organ pipe, flute, and recorder.

(b) *Reeds*, often made from actual reed or bamboo, are used in some organ pipes and in the clarinet, oboe, etc. Single and double reeds are possible, the former consisting of a thin wafer sharpened at the blowing edge, and the latter comprising two flat pieces bound round a small metal tube at the pipe end. In each case, the air stream is interrupted by the vibrations of the reed, so that the associated air column is triggered into resonance. When a reed is blown in free air, a note is emitted, rich in harmonics. When a resonant air column is coupled in, the natural frequencies of the column modify the reed quality, and dictate the pitch of the note.

(c) *The Player's Lips* are used to excite the air column into vibration in musical instruments of the brass family. The action resembles that of a double reed, and the natural rate of interruption of the air stream will be the resonant frequency of the lips and associated acoustical system. Usually this will be dictated by the air column, but some control is possible by changing the lip tension, and blowing pressure. Typical lip-reed instruments are the trumpet, trombone, and French horn.

2.8. A SHORT DESCRIPTION OF SOME STRING INSTRUMENTS (Fig. 2.4)

The Violin consists of a sound-box of special shape, closed except for two f-holes. The four strings are stretched over a bridge and

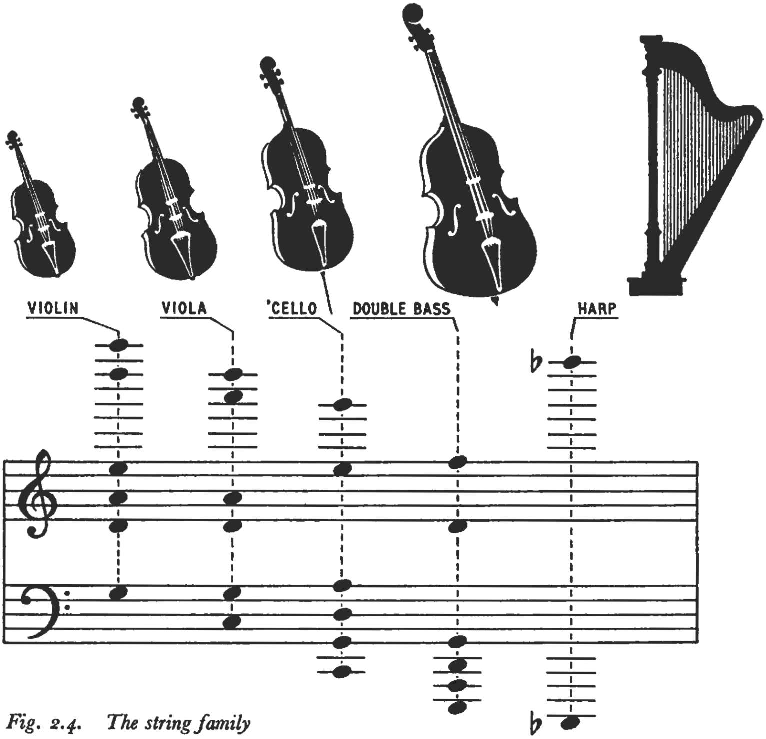


Fig. 2.4. The string family

tuned, by adjusting the pegs, to notes a fifth apart, as shown.

Stopping down with the fingers of the left hand gives a fundamental range of four octaves.

The Viola has heavier strings, whose open notes are tuned a fifth lower than those of the violin. Its fundamental range is over three octaves and its overall length is 2½ in. more than that of the violin.

The Violoncello or 'Cello is tuned an octave lower than the viola and has a fundamental range of over three octaves.

The Double Bass or Contrabass has heavy strings tuned in fourths, as shown. Its fundamental range is about three octaves, and it may be bowed or plucked (Fig. 2.4).

The Harp consists of 46 strings stretched on a triangular frame, one side of which is a small sound board. This sound board is a

relatively inefficient radiator, and the harp tone is characterised by the slow decay of sound. Seven pedals serve to sharpen the pitch of the notes—the C pedal controls all Cs and so on. The pitch shifting may be performed in two steps—a semitone or a full tone. The fundamental range is 6½ octaves, as shown.

2.9. A SHORT DESCRIPTION OF SOME WIND INSTRUMENTS (Figs. 2.5 and 2.6)

The Flute consists of a cylindrical tube which resonates as an open pipe. It is excited by means of a blowing hole for producing edge tones and is held in a transverse manner across the player's mouth.

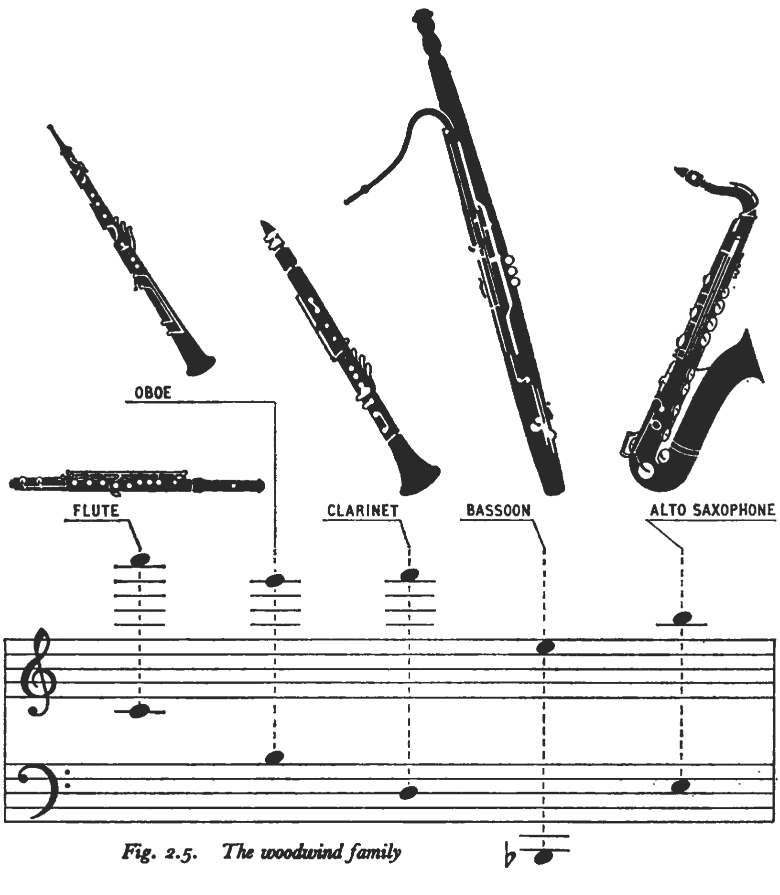


Fig. 2.5. The woodwind family

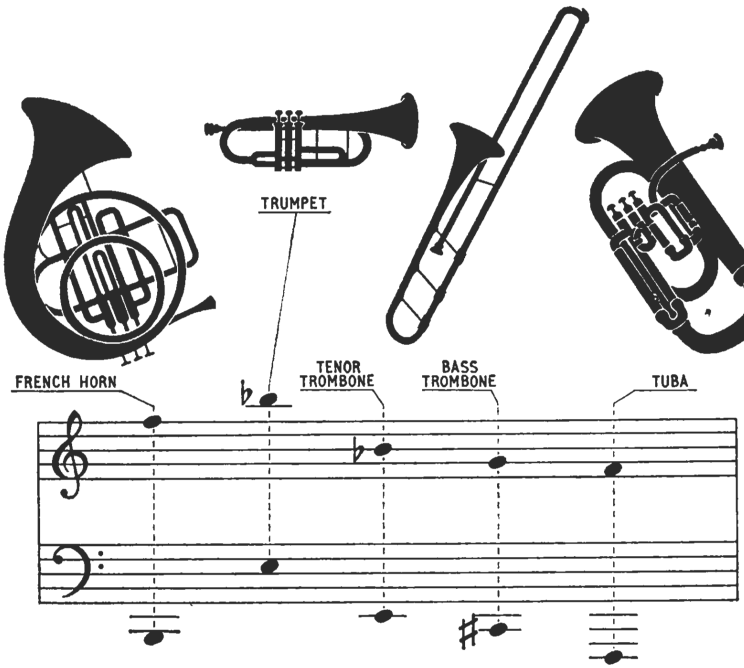


Fig. 2.6. The brass family

The blowing hole is situated a little distance from the actual end of the instrument, which is closed by a stopper, and a special open-end correction factor applies. The various notes of the scale are produced by uncovering holes along the pipe, thereby shifting the effective position of the open end. In early flutes it was necessary to make the holes of a convenient size, and a convenient position for the fingers. This imposed a limit on the variety and accuracy of notes. With the introduction of keys, and especially through the work of Theobald Boehm, great flexibility and trueness became possible. The modern flute has a fundamental range of three octaves from middle C.

Associated with the flute is a smaller transverse instrument called the *Piccolo*, which plays about an octave higher.

The Oboe is a double-reed instrument with a conical tube. Its fundamental wavelength is related to twice its length, and it is capable of producing the full series of harmonics. The oboe's note

A is often used for tuning purposes in orchestras, but a tuning-fork or electronic tone source at 440 c/s is more reliable. The fundamental range of the oboe is similar to that of the flute—three octaves from B.

Associated with the oboe is a larger instrument called the *English Horn*, or *Cor Anglais*, whose conical tube ends in a spherical bulb. It plays about a fifth lower than the oboe, and has a smaller compass.

The Bassoon is a double-reed instrument with a long conical tube (93 in. long) doubled back on itself for convenience. Its fundamental range is about three octaves, at a pitch two octaves lower than the oboe.

The Contra-Bassoon is folded three or four times, and plays one octave lower than the bassoon.

The Clarinet is a single-reed instrument with a cylindrical tube. Its fundamental wavelength is associated with four times its length, and its characteristic tone derives from the preponderance of odd-numbered harmonics produced. The system of keys is more complicated than in, say, the oboe, since over-blowing raises the pitch to the third harmonic (the nearest strong overtone), an interval of a twelfth. The fundamental range of the clarinet is over three octaves from D.

The Bass Clarinet covers a range one octave lower than the clarinet.

The Saxophone is a single-reed instrument with a conical tube employing (except in the soprano saxophone) a curved mouth-tube and upturned bell. The tube is wide compared with the clarinet, which makes for the more rapid attack or "speaking" of the saxophone. There are five members of the saxophone family—soprano, alto, tenor, baritone, and bass, of which the first and last named are rare. Each type of saxophone covers a fundamental range of about 2½ octaves, and the basic pitch falls in fifths from one type to the next. The fingered notes on the different saxophones are given the same names, the necessary shift in pitch being conveniently arranged by appropriate transposing from the written notes.

The French Horn consists of a tube about 12 ft long coiled on itself, and is a narrow cone with a wide bell. The instrument has a funnel-shaped mouthpiece to which the player applies his lips, varying their tension to produce the effect of a double reed. It is

possible to make the air column resonate, and play notes corresponding to about the first fifteen harmonic frequencies. Further notes, to cover the full chromatic scale, are produced by coupling in extra lengths of pipe. The earlier method of fitting a choice of U-shaped " crooks " to correspond to changes of key in the music gave some extension of the range. The modern system of three valves which introduce any combination of three extra lengths of tube has produced a flexible instrument with a fundamental frequency range of over three octaves.

By inserting his hand into the bell of the French horn the player is able to modify the effective length of the tube, in such a way as to raise or lower the pitch, or produce muted effects.

The Trumpet has a cup-shaped mouthpiece and wound tube, 6 ft in length, which is partly cylindrical and partly conical. Three valves, together with harmonic selection by the player, provide a chromatic scale over about three octaves. Metal or plastic mutes may be fitted into the flared bell to reduce the output and give a change in quality.

The Cornet looks like a more compact version of the trumpet, and covers a similar range of notes. Its tone is less brilliant, and it is a little easier to play.

The Trombone has a cup-shaped mouthpiece and a U-shaped cylindrical tube terminating in a flared bell. The tube is about nine feet long, doubled on itself, and possesses a telescopic section or slide, giving continuous variation of the resonant length of pipe. Sliding or glissando effects are therefore possible, and the intonation of notes depends on the player as in the violin, etc.

There are two sizes of trombone, the *tenor*, with a fundamental frequency range of about two and a half octaves, and the *bass*, with a range of about three octaves.

The Tuba is the largest orchestral brass instrument, and has an 18 ft coiled tube ending in a large bell pointing upwards. Three, or sometimes four, valves are used to give a fundamental frequency range of about three octaves.

2.10. PERCUSSION INSTRUMENTS (Fig. 2.7)

In all percussion instruments, the vibrations result from the instrument being struck by hammers, drumsticks, etc. The vibrating systems may be bars, plates, bells, or drumskins. It is possible

to divide the percussion family into two groups, depending on whether the sounds are of definite pitch or not.

2.10.1. Indefinite Pitch Instruments

In this category come the *triangle*, *bass drum*, *snare drum*, *cymbals*, and *gongs*. A very wide range of frequencies is generated by this group, including high frequencies up to and beyond the limit of audibility.

2.10.2. Definite Pitch Instruments

In this category come the *xylophone*, *glockenspiel*, *celeste*, *tubular and other bells*, and the *kettledrums* or *timpani*. A feature of these instruments is that the overtones are not usually simple multiples of the fundamental frequency.

The Pianoforte is usually listed as a percussion instrument. It consists of a keyboard, and a system of hammers which strike steel strings, stretched on a steel frame. A bridge couples the frame to a sound board which runs the whole length of the instrument. The fundamental range of the usual piano is over seven octaves—frequency range of 27·5 to 4,186 c/s.

Great variations in intensity are also possible (dynamic range)—so that this instrument puts the reproducing systems of broadcasting and recording to a severe test.

Depressing one of the keys flicks a hammer into momentary contact with the appropriate string(s), and at the same time withdraws the damping pad. When the key is released, the hammer

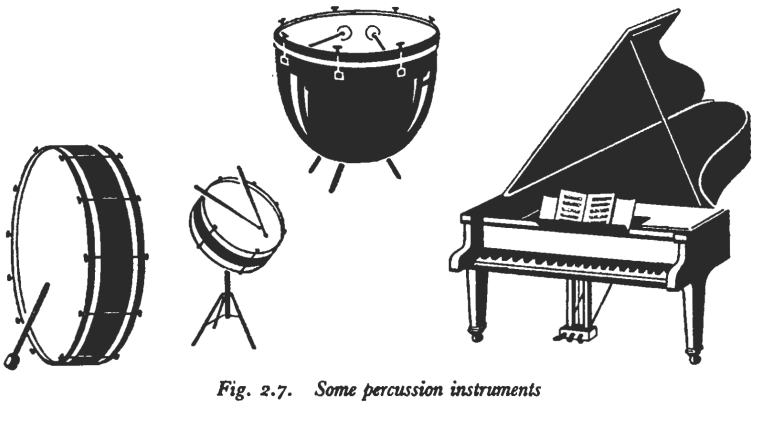


Fig. 2.7. Some percussion instruments

falls back into place, and the damper bears on the string and silences it.

Operating the right pedal, or *sustaining pedal*, removes all the dampers from the strings, so that the notes decay comparatively slowly. It also permits sympathetic vibration of strings which are harmonics of struck notes, causing considerable reinforcement of tone. A centre pedal is fitted in many pianos, called the *bass sustaining pedal*, which removes the dampers from the bass strings only. The left or *soft pedal* reduces the volume of sound in some way—by reducing the length of stroke of the hammers, or by shifting the hammers so that fewer strings are struck, or by keeping the dampers in contact with the strings.

2.11. THE HUMAN VOICE

The voice mechanism may roughly be described as a double-reed musical instrument. The *vocal chords*, when suitably tensioned, interrupt the steady flow of air forced between them by the lungs (Fig. 2.8).

The pitch of the fundamental note produced depends on the tension, thickness, and length of the vocal chords. All of these are capable of some variation in speech and singing, the basic length being greater for men than women. The male speaking voice has a

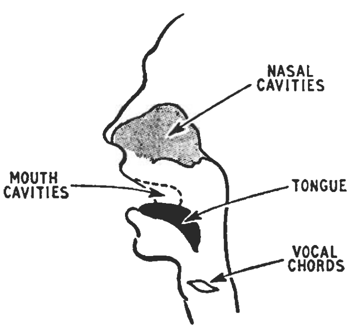


Fig. 2.8. The human voice

range of about 12 tones, centred on 145 c/s. The female speaking voice has an average frequency of 230 c/s with a similar compass.

The stream of air, pulsating according to the saw-toothed interruptions of the vocal chords, emerges into the cavities of the mouth and nose. It is the variation of the resonances of these cavities by manipulation of the lips and tongue which controls the harmonic

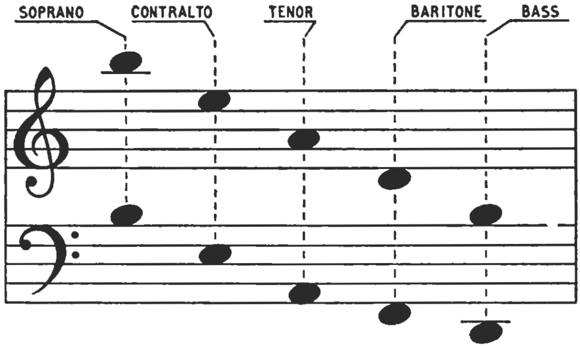


Fig. 2.9. Compass of singing voices

content of the sounds. Changes in vocal quality, not to mention emotional quality, correspond to variations in the relative strengths of partials. Investigations have shown that the production of each of the various vowel sounds depends on prominent resonances within two regions of pitch. These resonances, or more correctly bands of frequencies, are known as *formants*, and are usually specified by single frequencies. It must not be forgotten, however, that a formant frequency is in fact the centre of a wide band of sound. For example the vowel sound in the word " tone " is produced when frequencies in the region of 500 c/s and 850 c/s are stressed. The two formant frequencies for the vowel sound in the word " soon " are fairly low—400 c/s and 800 c/s—which explains the difficulty in singing this vowel clearly on high notes.

The fundamental frequency range of most singing voices is about two octaves, and average values are shown in Fig. 2.9.

3

STUDIO ACOUSTICS

In designing a theatre or concert hall for good acoustics, the following problems arise:—

(a) keeping out unwanted noise;

(b) obtaining adequate loudness at all points, without " dead spots ";

(c) obtaining good definition and intelligibility.

It will be seen that the last two requirements are in opposition, (b) calling for a gradual, and (c) for a rapid decay of sounds.

Broadcasting studios have a slightly different requirement, since directional microphones can be used to pick up sounds.

Television studios present special problems, due to the erection of a number of settings simultaneously, and to the high level of background noise often encountered.

3.1. SOUND INSULATION

An audience in a theatre will tolerate a level of extraneous noise which would be very distracting in a broadcast. This is because they are able to use both ears to locate the direction of the noise, and unconsciously ignore it, whereas the microphone does not discriminate against noise in this way. It follows that soundproofing is more important for studios than ordinary theatres, and Fig. 3.1 shows the minimum insulation required for speech studios. Insulation will normally be better than this, and in the case of music studios will be of the order of 70 dB.

It is usual to distinguish between:

(a) *air-borne noise*—entering through doors, windows, or ventilators, and including traffic noise, conversation and aeroplanes;

(b) *structure-borne noise*—due to impacts on the parts of the building, footfalls, machinery, slamming doors, and nearby studios.

In the case of air-borne noise, the reduction depends on the mass of the separating structures, and is most difficult to achieve at low frequencies. The reduction of structure-borne noise depends on discontinuous construction which forces sound to travel through a variety of different materials. Some reflection will take place at each

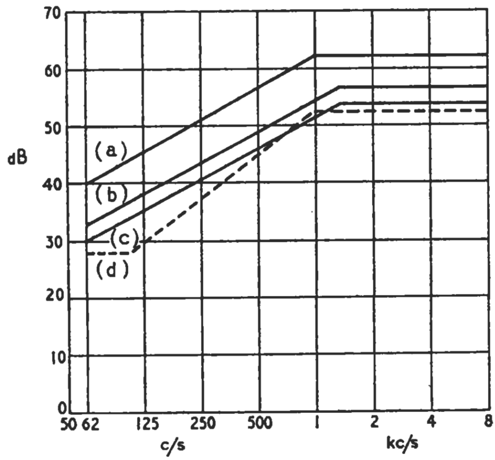


Fig. 3.1. Minimum figures for studio insulation

boundary, resulting in more or less rapid attenuation. In this case high frequency sounds are often well propagated and their attenuation can be a source of considerable difficulty.

3.1.1. Leakage between a Studio and its Cubicle

Poor insulation between a studio and its cubicle may give the studio manager a wrong impression of the quality actually being radiated. It frequently results in the impression of more bass than is really being sent to line.

Another danger, however, is the risk of "howl-back"—when a sound received by the microphone returns to the studio at an appreciable level via the cubicle loudspeaker. This condition will be set up if the gain from the microphone to loudspeaker exceeds the loss through the walls or doors, and the sounds from the monitoring loudspeaker will leak back into the studio and complete a circuit to the microphone (Fig. 3.2).

Sound insulation is usually less at low frequencies, as has already been mentioned, and the improved bass output of modern loudspeakers can give extra trouble. Howl-back will occur at some

frequency for which the path length is such as to return the sound to the microphone in phase with the original.

3.2. SOUND ABSORPTION AND REVERBERATION

An *echo* may be defined as a single reflection of sound following a noticeable time interval after the original. *Reverberation*, on the other hand, is a property of an enclosed space, and is the name given to the prolongation of sounds due to many reflections from the walls, etc.

When a sound commences in a studio, it does not immediately build up to its full intensity. The source is supplying energy at a given rate, but the waves are losing a certain percentage of their energy on each reflection from the boundaries. *Equilibrium intensity* is reached when the supply and withdrawal rates are equal. This " steady state " condition resembles the temperature of a room when heat from a radiator is being supplied at the same rate as it is lost through doors, etc. When the sound source ceases to vibrate, some time elapses before the reverberating energy is completely absorbed.

The diagram (Fig. 3.3), shows intensities reached by two pulses of sound under increasingly reverberant conditions. In the last

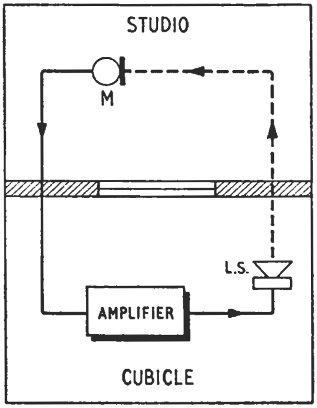


Fig. 3.2. Circuit round which "howl-back" may occur

drawing, the reverberation period is long, the build-up is considerable, and the intensity reached is high.

For a given sound, the equilibrium intensity, the time required for build-up, and for dying away, all depend on the " liveliness " or degree of reverberation in the studio.

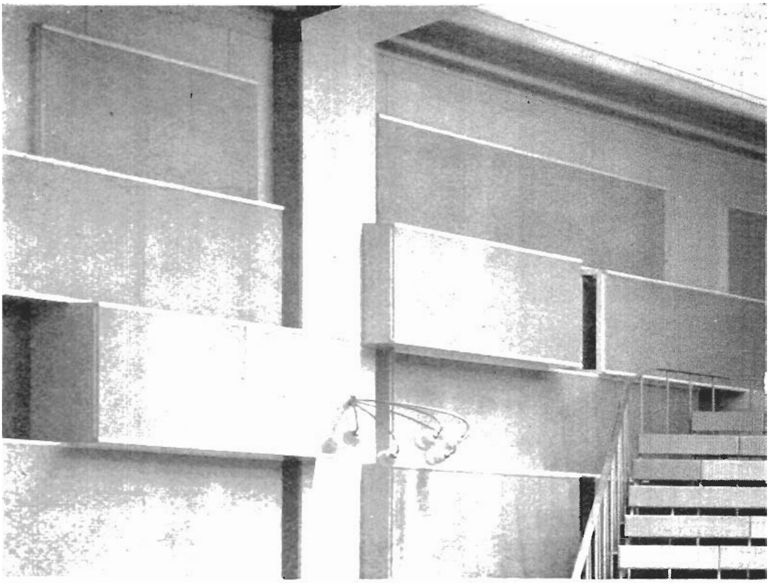


Plate 3.1. Part of Studio 1, Maida Vale, showing membrane absorbers



Plate 3.2. Part of Studio 1, Swansea, showing cavity absorbers



Plate 3.3. TV Studio, showing complexity of equipment

This in turn depends on the fraction of sound energy absorbed by the walls, etc. at each reflection and on the dimensions of the studio, which affect the time between reflections. W. C. Sabine was the first to attempt a precise analysis of these effects, varying one factor at a time. He defined the *reverberation time* of a room as

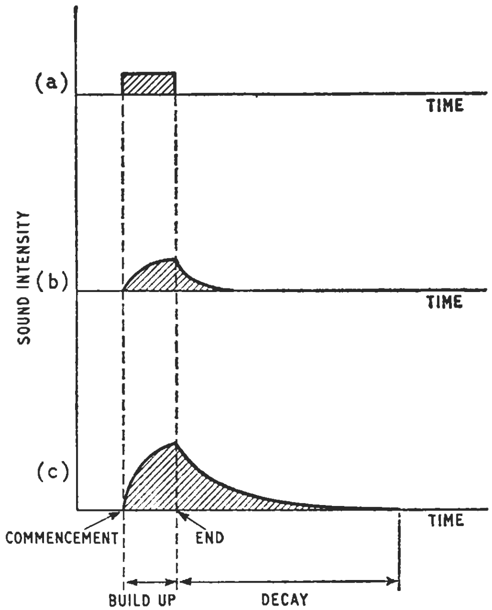


Fig. 3.3. Build-up and decay curves for enclosures with (a) short, (b) medium, and (c) long reverberation times

the time taken for a sound to fall to one-millionth of its intensity (through 60 dB). He also carried out two separate sets of experiments. In the first he measured the reverberation time in a number of bare rooms of different sizes, and found that the reverberation time was directly proportional to the volume.

In the second, he varied the amount of absorbent material in a given room, and found that the reverberation time was inversely proportional to the total absorption present.

From this it will be apparent that if all the available surface area in a studio is considered to be absorbing sound, the reverberation time will in fact be proportional to the cube root of the volume, since as the volume increases, so does the area of absorbing surface.

Only if the total absorption remains constant at all volumes does the reverberation time increase with the volume.

3.2.1. Optimum Reverberation Time

A certain amount of reverberation is desirable in a hall or a studio if adequate loudness is to be obtained at all points without straining the musician or speaker. Reflections which follow soon after the direct sound help to reinforce the volume of tone produced, hence the practice of using orchestral shells and other bright surfaces near concert platforms. But too long reverberation results in overlapping of syllables in speech with a consequent loss of intelligibility. In music, it is the definition which is lost, and the ability to distinguish the separate parts.

There is no law which will tell us how much reverberation will give the most acceptable compromise between these two requirements. But data has been collected with regard to certain halls judged by competent critics to be musically good. The results indicate that the reverberation time should rise nearly in proportion to the volume (more reinforcement being necessary to obtain adequate loudness) and that longer reverberation times are pre-

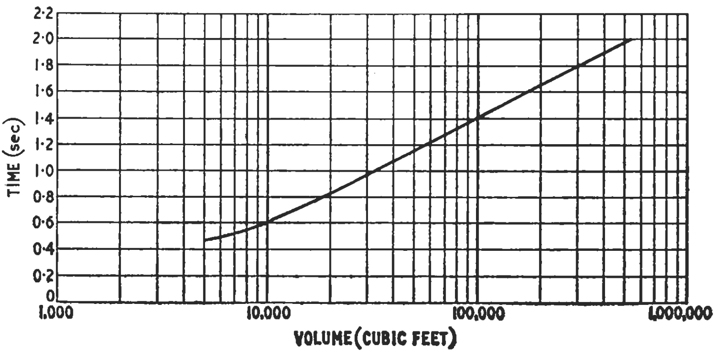


Fig. 3.4. Optimum reverberation time for enclosures of different volume

ferred for choral and orchestral music than for speech. This information has been conveniently summarised for music in the form of a graph (Fig. 3.4). For speech a time of 0·3–0·4 seconds is usually considered satisfactory.

It might be supposed that, in a broadcasting system, some allowance might need to be made for the fact that the listener's room will have a reverberation time which will be added to that of

the studio. However, in practice this is not usually a difficulty, and may be neglected if the studio reverberation time is long in comparison to that of the listener's room.

Some average reverberation times for various BBC studios and public buildings are given in Table 3.1.

Table 3.1

REVERBERATION TIMES OF STUDIOS, AUDITORIA, ETC.

The figures in the right-hand column represent the average reverberation time between 500 and 1,000 c/s, generally to two significant figures. This frequency range largely determines the subjective liveness. The studio figures allow for a typical number of performers, while those for concert halls and opera houses represent nearly full-house audiences.

Studio	*Purpose*	*Rev. Time (sec)*
Maida Vale, Studio 1	Symphony Orchestra	1·8
Maida Vale, Studio 2	Small Combinations	1·6
Glasgow, Studio 1	Scottish Orchestra	1·6
Birmingham, Studio 6	Midland Light Orchestra	1·6
Concert Hall, Broadcasting House	Small Orchestras, Chamber	1·4
Belfast, Studio 1	Northern Ireland Light Orchestra, etc.	1·4
Farringdon Hall	Music	1·7
Swansea Studio	Music	1·5
Camden Theatre	Light Music	1·5
Aeolian Studio 1	Light Entertainment	1·0
The Paris Studio, London	Light Entertainment	0·83
Cardiff, Charles St.	Welsh Orchestra	1·5
Studio 6A, Broadcasting House	Drama { Live End	0·68
	Drama { Dead End	0·17
Discussion Studios		0·3–0·4
Talks Studios		0·25–0·35
Concert Halls, Opera Houses, etc.		
Royal Albert Hall		2·9
St. Andrew's Hall, Glasgow		1·9
Royal Festival Hall		1·6
Usher Hall, Edinburgh		1·7
Free Trade Hall, Manchester		1·6
Colston Hall, Bristol		1·8
Concertgebouw, Amsterdam		2·0
Covent Garden Theatre		1·2
Glyndebourne Opera House		1·1
Wagner Theatre, Bayreuth		2·0
King's College Chapel, Cambridge	(approx.)	6·0

3.3. CONTROL OF REVERBERATION

The acoustic design of studios is complicated by the fact that no material will absorb equally at all frequencies. This means that more than one form of treatment will usually be necessary, and the

standard approach in BBC studios in the past has been to include porous absorbers and vibrating panel or membrane absorbers for the absorption of high and low frequencies respectively.

3.3.1. Porous Absorbers

A porous material such as cotton wool is an efficient absorber at middle and high frequencies. As the pressure fluctuates in the sound waves, air particles vibrate in the pores of the material, and dissipate energy in overcoming friction. For porous materials to be effective sound absorbers they must have a thickness approximately proportional to the wavelength of the sound to be absorbed. Hence for practical absorbers the absorption will be less at low frequencies. This can be improved by spacing the material from the wall, so

Table 3.2

VARIATION IN ABSORPTION COEFFICIENT FOR VARIOUS MATERIALS

Material	*Absorption Coefficient*		
	125 *c/s*	500 *c/s*	4000 *c/s*
1. Typical acoustic tile	0·19	0·68	0·85
2. 1 in. mineral wool with fabric cover	0·14	0·65	0·70
3. As 2, but with 1 in. air space behind	0·16	0·70	0·57
4. As 3, but with 5% perforated hardboard cover in place of fabric	0·17	0·96	0·25
Panel Absorbers			
Plywood 2 in. air space	0·25	0·20	0·10
Plywood air space stuffed with wadding	0·70	0·25	0·12
Hardboard and roofing-felt stuck together. 4¾ in. air space.	0·79	0·48	0·25
Membrane absorbers		See Fig. 3.6.	

that reflection from the wall surface behind the absorber can take place, thereby increasing the effective depth.

Table 3.2 shows the variation in absorption coefficient (i.e. fraction of sound absorbed) for various materials at different frequencies. It will be seen that a wide range of values is available.

3.3.2. Vibrating Panel Absorbers

Panels of plywood or hardboard supported about 2 inches out from the wall by battens one foot apart are found to possess natural frequencies in the range from 80 to 200 c/s. They are set in vibration by sound waves, and will take appreciable energy from the waves at and near their resonant frequency.

If the panel vibrations are damped, this energy is used up in overcoming friction, and the reflected wave is reduced in amplitude,

causing a reduction in reverberation.* In modern studio treatment, very efficient absorption together with pleasing decoration is achieved using either membrane absorbers, or cavity absorbers.

3.3.3. Membrane Absorbers

These selective low frequency absorbers consist of ordinary roofing felt stretched over the front of rectangular boxes (Figs.

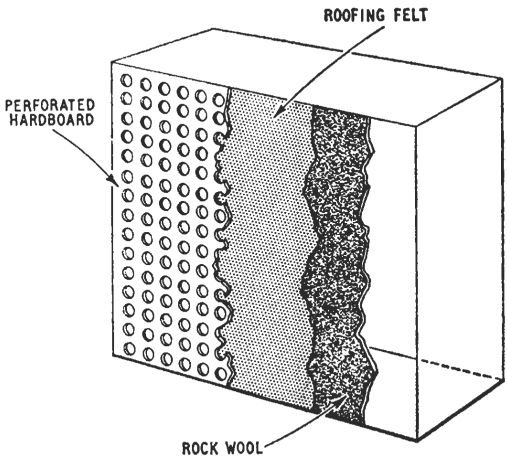


Fig. 3.5. Construction of membrane absorber

3.5 and 3.6). They have the following advantages over panel absorbers:—

(1) very efficient absorption—up to 100%;

(2) bandwidth is about half an octave—permitting selective treatment where necessary;

(3) absence of coloration due to re-radiation;

(4) they are cheap and easy to construct—and yet applicable to large and small studios.

The resonant frequency of the unit at which absorption is a maximum depends almost entirely on the depth of the box.

Most of the absorption is due to the internal friction of the felt, and a blanket of rock wool may be inserted where additional

* A lightly damped structure tends to continue sounding for a long time, and re-radiate the sound. This kind of coloration has sometimes caused trouble in studios. The "ringing" of lampshades or music-stands is an example of this effect.

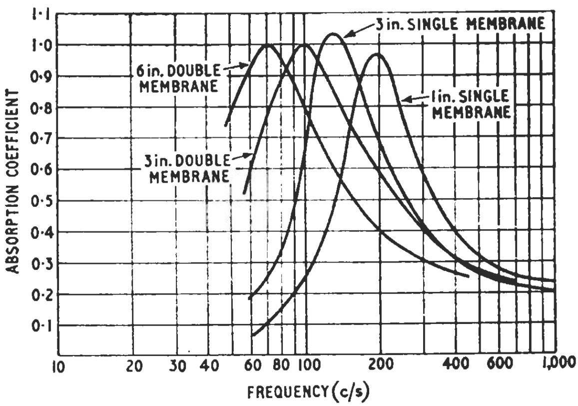


Fig. 3.6. Absorption coefficients of roofing felt membrane absorber

damping is required—to widen the band of frequencies absorbed. When high frequency absorption is also required, the protective covering is made of canvas or silk.

To prevent sideways vibration of the air, the cavity is partitioned at spacings less than a wavelength.

3.3.4. Restriction of Absorption at High Frequencies

High absorption coefficients at high frequencies may well be undesirable. The furnishings and occupants of the studio tend to absorb high frequencies and so additional absorption might well make the studio too dead. The high frequency absorption of the units described can be restricted by covering them with perforated hardboard, which will reflect high frequencies.

e.g. 5% holed hardboard reflects above 1,000 c/s

20% slotted hardboard reflects above 4,000 c/s

In a particular studio treatment, four depths of resonant unit, tuned to 62, 80, 250, and 300 c/s are used in Studio One, Maida Vale (Plate 3.1). Since this studio was previously bass heavy, little or no high frequency treatment was included. The projecting units also serve to break up the sound waves, and give better diffusion. If this is not required, the units may be recessed, or fitted along the wall/roof angle.

3.3.5. Cavity Absorbers

These make use of a principle first expounded by Helmholtz—namely, the resonating properties of air in a narrow-necked container. The mass of air in the neck of a bottle, for example, will vibrate against the "spring" of the enclosed air if an "edge tone" is produced by blowing across the top of the bottle.

Provided the resonance of cavity absorbers is damped, reverberation will be reduced, and different sizes of absorber may be used throughout the audio frequency band. For example, in Denmark a particular dance music studio uses three sizes of cavity absorbers giving a reverberation time of approximately 0·8 seconds at all frequencies. In this country, cavity absorbers consisting of rows of cardboard tubes let into plaster boxes have been used successfully for low frequency absorption. In Swansea Studio One, line arrays of eight cavities are fixed to the walls to form an unusual decorative effect (Plate 3.2).

3.4. ROOM RESONANCES

When sound waves are reflected and re-reflected between parallel walls, the amplitude of the resulting standing wave is a maximum

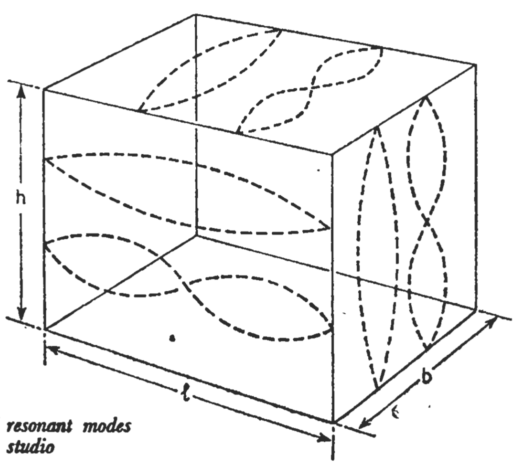


Fig. 3.7. Representation of resonant modes in a rectangular studio

when the walls are a multiple of half a wavelength apart. Thus, in a rectangular studio, a harmonic series of resonant frequencies exists related to the length, breadth, and height (Fig. 3.7). These resonances are called *eigentones*, and are undesirable, since

they cause coloration of the studio output. They are most pronounced in small studios, where the first few harmonics lie within the band of speech frequencies.

In some Talks studios, the resonances are so marked that they cannot be eliminated directly by means of absorbers. It is then necessary to use a *Microphone Correction Unit* (see Fig. 4.14).

This has the disadvantage that the direct sound also is " corrected " while the " boominess " is still a distraction to the speaker in the studio.

3.5. DIFFUSION OF SOUNDS

So far, we have assumed that the sound energy is distributed evenly in the studio—but this is not usually the case. Standing waves are set up near walls, sounds are focused in certain spots by concave surfaces, and frequency distributions are disturbed by selective reflections from acoustic materials. For example, the recent attempts at " directed sounds "—using orchestral shells, and a fan-shaped hall—have tended to give a hardness of tone, presumably due to the strong direct sound and the uneven decay.

A more diffuse sound is obtained by arranging convex or triangular splays or spherical surfaces. There should still be a region of strong reflection near the orchestra, for best ensemble playing, and such reflectors should preferably be overhead or at the sides, parallel to one another.

A live- and dead-end studio, where all the acoustic treatment is arranged at one end, gives a useful variety of reverberation conditions within one set of four walls—e.g. Studio 6A, Broadcasting House (see also Chapter 13).

3.6. EFFECT OF MICROPHONE DISTANCE

In all except the very largest studios, the average intensity of reflected sound is roughly equal at all points. Thus, for a given amplifier setting, the indirect sound reaching the microphone—and therefore the loudspeaker—is the same for all microphone positions; see shaded region at (a) in Fig. 3.8.

Reducing the microphone distance increases the volume of the direct sound (b), and if we restore the original volume by fading down on the amplifier (c) we reduce in proportion the amount of reverberant sound heard by the listener.

In a given studio, then, it is possible to get a range of " apparent acoustics " by suitable choice of microphone distance. This fact

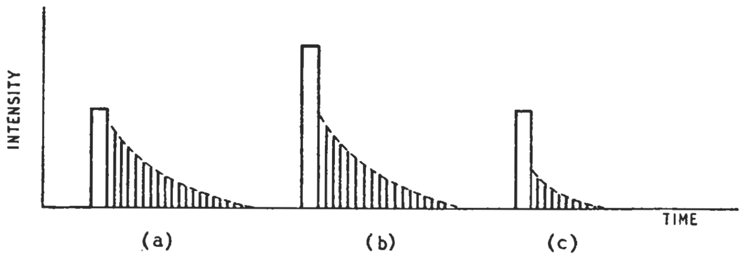


Fig. 3.8. Reduction in "apparent reverberation" by use of closer microphone

has many useful applications. It has the disadvantage, however, that a widely-spread source—such as an orchestra—may present several aural perspectives at once, unless the layout is carefully arranged to avoid this.

Also it restricts cast movements in broadcast plays—the increase in apparent reverberation due to stepping back a few paces may sound like moving into the middle distance.

3.7. SPECIAL PROBLEMS OF TELEVISION STUDIOS

For a variety of reasons, a television studio cannot be built on the same lines as a studio used for sound only. In particular, the acoustic problems are quite different. Even the smallest television

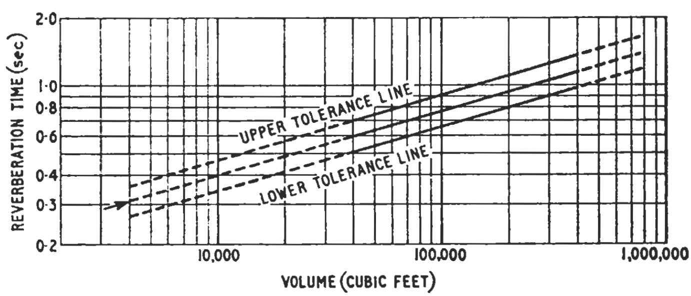


Fig. 3.9. Optimum reverberation time for General Purpose TV Studios

studios are large by sound standards, in order to accommodate banks of lights, cameras and crews, microphone booms, special effects units, and scenery associated with a number of sets (see Plate 3.3). To bring the reverberation time of such a large enclosure down to the required value for Talks and Drama necessitates heavy

acoustic treatment. It is not unusual for almost the entire wall area to be covered with glass wool, held in position by perforated board or chicken wire. Roofing felt units are also employed. The curve in Fig. 3.9 shows the variation in reverberation time with volume found to be most satisfactory.

A further reason for making television studios comparatively " dead " is the need to keep the microphone " out of shot ", which often makes the working distance greater than that used in a comparable sound broadcast. As actors, variety performers, etc. move about, they are followed in many cases by the camera, and also the microphone. A boom is used, carrying a directional microphone on a telescopic arm. It is the job of the boom operator to swing the microphone to and fro, and to rack it in and out; at the same time, by suitable manipulation of a hand swivel, keeping it pointing at the actors. This calls for a great deal of skill and experience, and is beset by three difficulties:—

(a) keeping the microphone " out of shot ";

(b) keeping the shadow of the microphone " out of shot ";

(c) relating the aural perspective to the visual perspective.

(a) and (b) are mainly a question of experience, the operator just avoiding the imaginary line joining the lens and the top edge of the televised scene. If a complicated lighting arrangement is used, microphone movement may be severely limited, and the two facilities should be planned together when possible. (c) is a constant problem, although the ear is extremly tolerant of anomalies of perspective when the eye is also in use. It is possible to line up the correct apparent perspective for ear and eye in a given shot—say a close-up. If the camera now moves back, or switches lenses, or a change is made to a more distant camera, the boom operator must move the microphone above the rising lens boundary, and take up a position which " sounds " at the new distance of the picture. This operation is complicated, and not one in which any rules can apply, especially when one remembers that in the monophonic* system of sound reproduction distances are not what they seem.

* The increasing use of " stereophonic " as a term to describe a two-channe system so arranged as to produce a spatial distribution of sound has necessitated a new term for single-channel recordings and transmissions. The term " monaural " scarcely fills the bill, since most people listen to one loudspeaker with two ears. The term would be strictly true only of single headphone listening. The term " monophonic ", signifying one sound source (i.e. one loudspeaker) would appear to be more suitable and is coming into general use.

Before leaving the subject of television studio acoustics, the effect of the actual sets themselves must be mentioned. Panelled flats and the like in a room setting will mask the acoustics of the studio proper. The setting up of marked reflections or standing waves is even more undesirable when the microphone is moved during the action, and canvas scenery which is transparent to sound is to be preferred.

4

BASIC ELECTRICAL THEORY

4.1. ATOMIC NATURE OF ALL MATERIALS

All the thousands of chemical and everyday substances which exist have been found to consist of different combinations of a comparatively small number of basic materials known as *elements*. The gases hydrogen and oxygen, for example, are elements. Ordinary water is found to be not an element but a combination, in certain proportions, of these two gases (the ubiquitous H_2O). There are about a hundred different elements, and science has discovered here a further simplification in the grand scheme of nature. All

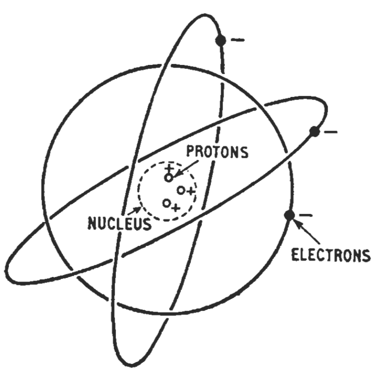


Fig. 4.1. Representation of atom possessing three electrons

the various elements are made up from combinations of a mere handful of basic units, which are recognised only by their weight and electrical charge.

The smallest particle of a given element is called an *atom*. Atoms are in the news nowadays, and most people know that the atom is constructed like a tiny solar system, with a heavy *nucleus* round which lighter particles circle in orbits, like planets round the sun (Fig. 4.1). The difference between one element and another is

simply based on the numbers of planetary and nuclear particles which go to make up their atoms. The planetary particles are called *electrons* and the principal nuclear particles are called *protons*. Hydrogen, the simplest element, has atoms in which one electron circulates round a single proton. The electron is prevented from flying away from the proton by a force of attraction existing between them, similar to the force of gravity which keeps the moon in orbit round the earth. The force which keeps the electron in its orbit is electrical. The electron and proton are said to carry equal *negative* and *positive* charges of electricity respectively, which produce a force of attraction between them. Like charges, on the other hand, are found to repel each other. Normally, the total numbers of electrons and protons are equal, the electrical charges cancel, and everyday substances show no overall charge.

4.2. ELECTRIC CURRENT

In some materials the electrons and protons are held together very strongly and it is difficult for any electrons to be detached from their atoms. These materials are called *insulators* and include glass, rubber and bakelite.

In other materials, notably carbon and most metals, the force of attraction between the protons and the electrons in the outer orbits is small enough for these electrons to be easily detached from their parent atoms. It is these free electrons moving within the material that can constitute an *electric current*. The force needed to maintain this current is known as the *electromotive force* (e.m.f.) and is measured in *volts*. The e.m.f. can be produced by a battery, which works by electro-chemical action, or by the effect of a magnetic field, as will be seen later in the chapter.

The flow of current is measured in *amperes* (or *amps*), units which correspond to a certain number of electrons per second passing a given point in the circuit. This may be compared with measuring water flow in a pipe in gallons per second. As an indication of the minute scale of quantities in the atom, the number of electrons per second corresponding to one amp is:

$$6 \times 10^{18} \text{ or } 6{,}000{,}000{,}000{,}000{,}000{,}000$$

It may be noticed that the conventional direction of current flow is from the positive (red) terminal to the negative (black) terminal, while the actual direction of the electron stream is opposite to this, because the electron is a negatively charged particle.

4.3. RESISTANCE AND OHM'S LAW

The question now arises—how many amps of current will flow in a given circuit for the application of a given number of volts? Circuits vary a great deal, and a thin wire of copper, for example, offers more " resistance " to current than a thick wire. Ohm experimented extensively on this phenomenon and found that in a given circuit the current is proportional to the e.m.f., so that doubling the volts doubles the amps. This proportionality is called *Ohm's Law*, and may be written:—

$$\frac{V}{I} = R$$

where V is the e.m.f. in volts, I is the current in amps, and R is a constant for the circuit. Ohm called this the *resistance* of the circuit. The unit in which resistance is expressed is the *ohm* which may be defined as follows:—A circuit has a resistance of 1 ohm if an e.m.f. of 1 volt produces a current of 1 ampere. Resistance might also be defined as the ratio of voltage to current.

4.3.1. Examples of Ohm's Law

Example 1

The e.m.f. applied to a certain circuit is 12 volts, and the current is 3 amps. What is the resistance in ohms?

$$V/I = R$$

$$\therefore \quad R = 12 \div 3$$

$$= 4 \text{ ohms}$$

Example 2

How many volts must be applied to a circuit of 50 ohms to produce a current of 3 amps?

$$V/I = R$$

$$\therefore \quad V = IR$$

$$= 3 \times 50$$

$$= 150 \text{ volts}$$

4.3.2. Variable Resistors

We have seen that the current resulting from a given voltage depends on the resistance in the circuit. Thus it is possible to use

resistors as circuit components to control the current. These resistors can be made from special resistance wire, whose resistance is known to be " so many ohms per foot ", or from blocks containing powdered carbon.

In Fig. 4.2, for example, the current will be 6 amps (12 ÷ 2), or 4 amps (12 ÷ 3), depending on whether the switch is thrown to the left contact or the right.

If two resistors are connected *in series*—i.e., end to end—the same current will flow through both, and its value will be determined by

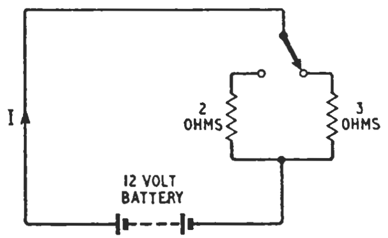


Fig. 4.2. Resistance

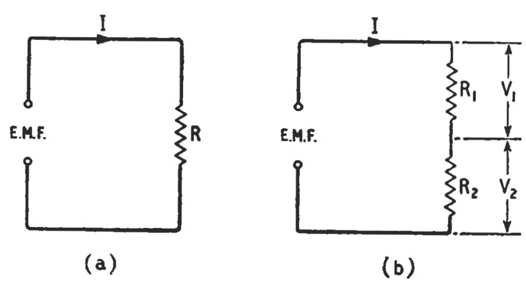


Fig. 4.3. Resistors in series

the sum of the two resistances. In Fig. 4.3, circuit (a) is equivalent to circuit (b) when

$$R = R_1 + R_2$$

Now applying Ohm's Law to the series circuit, we have three voltages:—

$$V = I \times (R_1 + R_2)$$

$$V_1 = I \times R_1$$

$$V_2 = I \times R_2$$

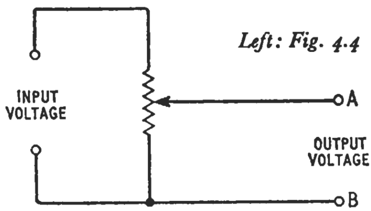
and the applied voltage V is seen to be divided into two parts,

called the " voltage drop " across R_1 and R_2. The ratio into which V has been divided is the ratio of the two resistors.

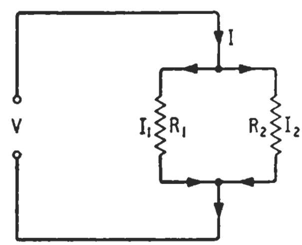
This is the basic circuit of a *potential divider*, and if a sliding contact is provided to produce continuous variation of the voltage tapped off across A B, we have a potentiometer or fader (see Fig. 4.4).

4.3.3. Resistors in Parallel

If the ends of two resistors are connected together (Fig. 4.5), and a voltage is applied to both, the current will divide into two parts.



Left: Fig. 4.4 Variable resistors



Right: Fig. 4.5. Resistors in parallel

The currents through R_1 and R_2 are found from Ohm's Law to be

$$I_1 = V \div R_1$$

$$I_2 = V \div R_2$$

$\therefore$ the current is divided in the ratio

$$\frac{I_1}{I_2} = \frac{V}{R_1} \times \frac{R_2}{V}$$

$$= \frac{R_2}{R_1}$$

which is the inverse ratio of the resistances.

Note:

(a) This is what we should expect—more current will pass through the smaller of the two resistances—and a " current divider " has been constructed.

(b) The effective resistance of R_1 and R_2 in parallel is not found from the formula $R = R_1 + R_2$ as in the series case, but from

$$\frac{1}{R} = \frac{1}{R_1} + \frac{1}{R_2}$$

4.4. POWER IN ELECTRICAL CIRCUITS

A battery is doing work when it drives a current round a circuit, and the rate of doing work is called the *power*. The unit of power is the *watt*, and the number of watts being consumed in a circuit driven by a voltage V is given by the formula:—

$$W = V I$$

By Ohm's Law, replacing V by IR,

$$W = I \times IR$$

$$= I^2 R$$

Similarly by replacing I,

$$W = \frac{V^2}{R}$$

This shows us that increasing either V or I increases the power.

Note that although we talk of the power being expended, or used up, it is a necessary consequence of the Law of Conservation of Energy that the electrical energy must reappear in some other form —as heat, light, or motion.

4.4.1. Examples on Electrical Power

Example 1

An electric fire has a coiled wire element whose resistance is 40 ohms. What is the power dissipated when the fire is connected to 200-volt mains?

$$W = V^2R$$

$$= \frac{200 \times 200}{40} \times \frac{40{,}000}{40}$$

$$= 1{,}000 \text{ watts}$$

Example 2

What current is taken by a 100-watt electric lamp connected to 240-volt mains?

$$W = VI$$

$$\text{i.e. } I = \frac{W}{V}$$

$$= 100 \div 250$$

$$= 0{\cdot}4 \text{ amps}$$

4.5. CAPACITORS AND THE STORAGE OF ELECTRICITY

Imagine that the two lines at C in Fig. 4.6 are a pair of metal plates with an air space between them. Close the switch S.1 and electrons will tend to flow round the left-hand circuit. They will

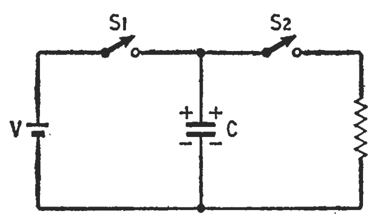


Fig. 4.6. Storing electrons in a capacitor

be attracted to the positive terminal of the battery, and repelled from the negative terminal. Since no current can flow across the air space (air is an insulator), a condition of strain is set up, with an excess of electrons collected on the lower plate. The plates are said to form a condenser or capacitor, and if switch S.1 is now opened again, the charge on the two plates is held in suspension.

Closing S.2 completes the right-hand circuit, and electrons will flow through R until the normal uncharged condition is restored.

The storing ability of a capacitor is called its *capacitance* (C), which is measured in Farads. A common practical unit is the micro-farad, μF; 1 farad = 1,000,000 μF. A capacitor has a capacitance of 1 farad if a pressure of 1 volt sets up a charge of 6×10^{18} electrons (the same number as flow per second in a current of 1 ampere).

The capacitance of a given pair of plates is found to be directly proportional to their area, and inversely proportional to the

distance between them. If the insulator between the plates is mica or paper instead of air, a larger capacitance results. The " capsule " of a condenser microphone is an example of a capacitor, with the diaphragm forming one plate and the fixed plate of the microphone the other (see Chapter 5).

4.6. MAGNETISM AND INDUCTANCE

A charged capacitor, with its positive and negative plates, has been considered as a means of storing electrical energy. A magnet, with its North and South poles, stores energy in the form of a magnetic field. This energy can be put to use, attracting pieces of iron or steel, just as a charged capacitor can drive electrons round a circuit.

The process of magnetising a piece of steel is one of aligning groups of molecules, each group acting as a tiny magnet, so that they lie in line instead of in a random arrangement. These groups of molecules are termed *domains*. The domains of iron are comparatively easily turned, whereas hard steel, once magnetised, retains its magnetic qualities for a long time (Fig. 4.7).

When current is passing along a wire, a magnetic field is produced, and if the wire is wound into a coil, the magnetic field is identical to that which surrounds a bar magnet. The polarity—i.e., position of the North and South poles—depends on the direction of flow of the current (Fig. 4.8). The coil in a tape recording head, for example, is fed with alternating current so that the " polarity " of the poles is continually being reversed, twice in each cycle.

Now consider the circuit in Fig. 4.9, with a coil L and a resistor R, in series. When the switch is made, current will begin to flow, but will not immediately be at the rate decided by R alone, since the magnetic field is being built up—which requires energy. Once L is fully " charged ", so to speak, no further energy is required to keep it in its magnetic state, and $I = V/R$, as Ohm's Law would predict. In the same way, when the voltage is removed, the collapse of the magnetic field tends, for an instant, to produce a voltage to maintain the current, thus paying back the energy " borrowed ". In this way a coil of wire resists *changes* in the current, and this property is called *inductance*. The unit of inductance is the *henry*.

If a core of iron is fitted inside the coil, the magnetic field is much strengthened, and we have an *electromagnet*.

The collapse of the magnetic field in the example above can be thought of as a " motion " of the field relative to the coil, and this motion produces the voltage in the coil. In similar fashion, if a

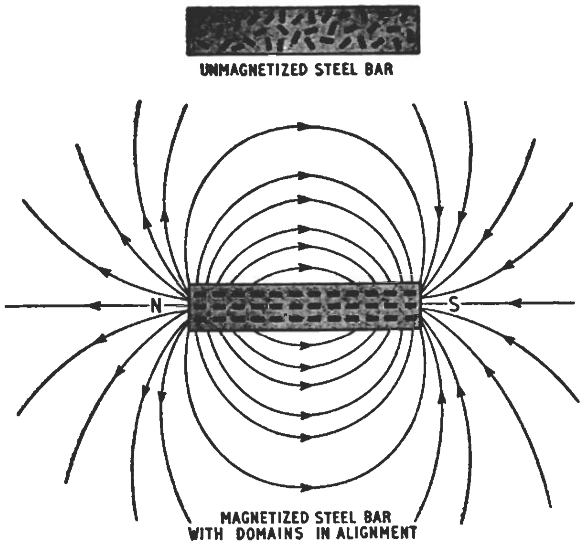


Fig. 4.7. Magnetic field surrounding a bar magnet

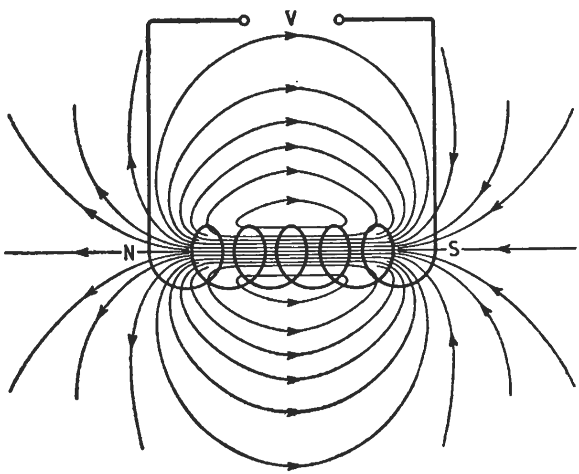


Fig. 4.8. Magnetic field surrounding an electric current

permanent magnet is moved inside a coil of wire, thereby creating a moving magnetic field, a voltage is induced in the wire so long as the magnet is moving. A voltage difference will therefore exist at the terminals of the coil. This is the principle of the simplest form of electrical current generator.

4.7. USES OF ELECTROMAGNETISM

4.7.1. Relays

A *relay* makes switching at a distance possible. Applying a small current to the electromagnet in Fig. 4.10 energises it, so that it attracts the iron lever controlling the switch. Thus the cue lamp

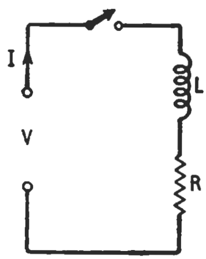


Fig. 4.9. Inductance

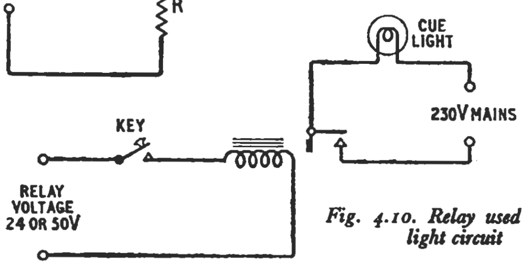


Fig. 4.10. Relay used in cue light circuit

circuit is completed, and the lamp will light. By extension of the relay circuit, there is almost no limit to the distance at which the circuit will operate. The 100 kw Third Programme transmitter at Daventry for example, is controlled by relays operated in the main control room some considerable distance away, and this control circuit has in fact been extended in the past so that the transmitter could be switched on in the London control room.

4.7.2. Buzzers

Buzzers and electric bells work on the same principle, with the addition that positive action of the attracted lever is made to break the relay energising current, so that the lever falls back, and

the action is repeated for as long as the buzzer switch is depressed (Fig. 4.11).

4.7.3. Transformers

In a transformer there are two coils of wire wound on a core, electrically insulated from each other. In a special type of transformer, the auto-transformer, the two windings are connected in series, but the operation is the same. It is convenient in drawings

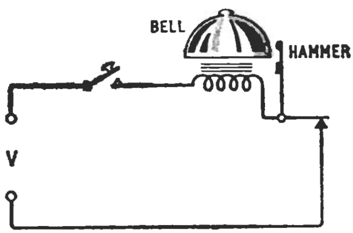
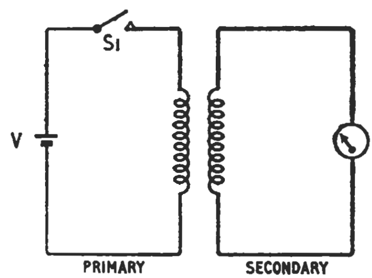


Fig. 4.11 Electric bell

to show the coils separately. We have seen that at the moment of switching on or off, the build-up or collapse of the magnetic field surrounding a coil tends to oppose or reinforce the current.

In a transformer, changes in the magnetic field associated with current in the *primary* winding induce an e.m.f. in the *secondary*.

Fig. 4.12. The transformer



In Fig. 4.12, for example, the current meter in the secondary winding will give a momentary reading first in one direction and then in the other, as the switch S.1 is closed and then opened.

A special type of transformer with two secondary windings is sometimes used in studio circuits. It is called a *hybrid transformer*, and it divides the input into equal and independent halves in such a manner that no interaction can take place between them. With

a normal transformer, any signal fed into one of the secondary windings would appear at the other and the hybrid is so designed that this cannot occur. (See Fig. 7.19.)

4.7.4. Loudspeakers

When a conductor carrying current is placed in a magnetic field, there is an interaction between the two magnetic fields—the static one and the one associated with the electric current. The general effect of this interaction is to produce a force tending to cause the conductor to move. The direction of movement is at right angles to both the current and the magnetic field, and it is reversed if either the current or field is reversed.

This principle is employed in the moving-coil loudspeaker. Electric currents which correspond to the original sound waves in frequency and amplitude are passed through a coil of wire. The coil is suspended in the field of a permanent magnet, specially shaped to produce a radial field, and is accordingly set in to and fro movement. These vibrations are taken up by the paper cone to which the coil is attached and sounds are radiated into the surrounding air. (See Loudspeakers, Chapter 6.)

4.7.5. The Moving-coil Microphone

The process of converting electrical energy into sound energy just described for a loudspeaker is reversed in the moving-coil microphone. Here a coil is suspended in the field of a magnet, as before, and attached to a light diaphragm capable of vibrating in response to sound waves. When the diaphragm—and therefore the coil—moves to and fro, an e.m.f. tends to be induced in the coil. If a circuit is completed, perhaps to an amplifier, a current is produced corresponding to the original sound vibrations. (See Microphones, Chapter 5.)

4.8. ALTERNATING CURRENTS

The electric currents discussed at the beginning of this chapter flowed in a certain direction round the circuit (plus to minus) as driven by a battery or similar electromotive force. These are called direct currents (d.c.). In broadcasting we are more often concerned with currents which are continually reversing their direction. These are called alternating currents (a.c.), and the number of double reversals (cycles) per second is called the *frequency*. This lines up with our treatment of sound-wave theory.

Free electrons in a wire carrying 1,000 c/s tone currents are performing simple harmonic motions identical to those performed by air particles in the sound wave from a 1,000 c/s tuning sound.

The moving-coil microphone described in section 4.7.5. is seen to be a generator of alternating currents. Driven by the sound wave, the coil moves in the field of the magnet, first in one direction and then in the other, giving rise to an e.m.f. which is continually reversing. Imagine that the graph of the fork vibrations is a sine wave. This wave form will apply to the coil vibrations also and therefore to the e.m.f., the voltage performing 1,000 swings positive and negative per second.

It is now proposed to summarise the behaviour of a.c. in circuits containing resistance, capacitance, and inductance.

4.8.1. A.C. and Resistance

Fig. 4.13(a) shows an a.c. generator supplying current to a circuit consisting of a resistance R.

The current is given by Ohm's Law ($V = IR$) as with d.c., and is the same at all frequencies. Voltage and current reach their maximum values together—i.e. they are *in phase*. To calculate the *power* in the circuit, peak or maximum values are not used: instead an average known as the root mean square value (r.m.s.) is

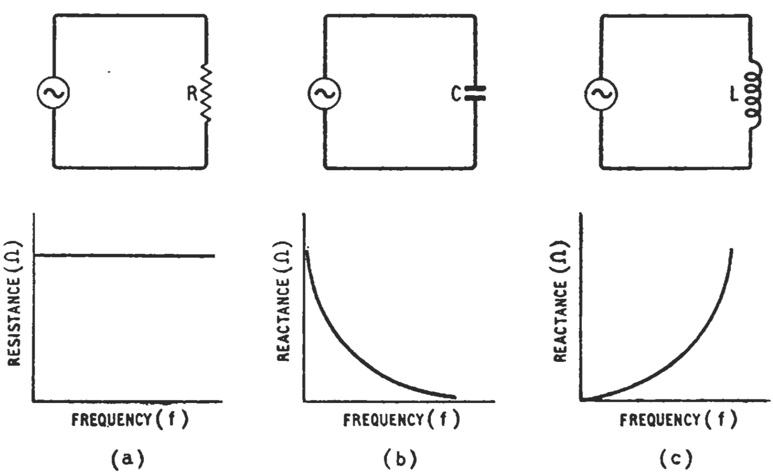


Fig. 4.13. Different opposition to the flow of a.c. of (a) resistance, (b) capacitance, and (c) inductance

taken and this is found to be approximately 0·7 times the a.c. peak. Thus, an a.c. power supply rated at 230 volts swings between + 325 volts and − 325 volts.

4.8.2. A.C. and Capacitance

Fig. 4.13(b) shows a.c. applied to a capacitor. The alternating e.m.f. causes an alternating charge to appear on the capacitor, with electrons accumulating first on one plate, and then on the other. Thus, although we have seen that d.c. will not flow through a capacitor, a.c. can be said to flow in and out of it. The magnitude of this current (number of electrons per second) is found to increase with frequency, and also with the capacitance. Thus the opposition of a capacitor to current flow is *inversely proportional to both frequency and capacitance.*

4.8.3. A.C. and Inductance

Fig. 4.13(c) shows a.c. applied to an inductor. We have seen that inductance opposes change in the current and might expect the opposition to increase with frequency. Such is indeed found to be the case. Increasing the size of the inductor also increases the opposition to a.c., and so this opposition in the case of an inductor is *proportional to both frequency and inductance.*

4.8.4. Reactance and Impedance

The property of opposition to alternating current flow in the capacitor and the inductance is known as *reactance* (X), in order to distinguish it from the similar property of a resistor known as resistance (R). In most cases, of course, in a given component, reactance and resistance will be present together—a coil of wire forming an inductance must have a small resistance—and when both are present, the combined effect of both resistance and reactance is known as *impedance* (Z) (e.g. the impedance of a loudspeaker is often quoted as 15 ohms where the measured resistance may be only of the order of 5 ohms: the difference is made up of reactance due to the inductance present).

4.8.5. A.C. and the Transformer

If an alternating current is applied to the primary winding of a transformer, each reversal of direction of the current will cause a change in the direction of the magnetic field in the core, and these changes in their turn will cause an alternating current in the

secondary winding of the transformer. The magnitude of the voltage measured across the secondary winding, due to this secondary current, depends upon the turns ratio between the primary and secondary. It is therefore possible, by suitably adjusting this ratio during manufacture, to arrange for the secondary voltage to be stepped up or down as required in relation to the voltage applied to the primary.

It might appear that when a secondary voltage higher than the primary voltage is required, " something for nothing " is obtained. This, of course, is impossible since the *power* taken from a transformer cannot be more than the power put in and so, when the voltage is increased, the current in the secondary winding is proportionately decreased.

$$\frac{V_1}{V_2} = \frac{T_1}{T_2} = \frac{I_2}{I_1}$$

where T_1 and T_2 are the number of primary and secondary turns.

A transformer may have several secondary windings where different ratios and voltages are required.

The transformer can also be used to convert from one impedance to another and the impedance ratio

$$\frac{Z_1}{Z_2} = \frac{T_1^2}{T_2^2}$$

4.9. OUTPUT IMPEDANCE

All sources of voltage, such as batteries, microphones and amplifiers possess internal resistance, and since reactance may be present also, this is given the general name *output impedance.* It is the impedance measured " looking back " from the output terminals. It is not proposed to discuss this difficult subject at any length, but one important aspect deserves mention—namely, that a source delivers maximum power into a circuit when the impedance in the circuit is equal to the output impedance of the source. The source and "load" are then said to be "matched"—compare the use of a matching transformer in microphone amplifiers to step up the low output impedance of the microphone to the considerably higher impedance of the first valve.

In devices which have only limited linearity, for instance, valve amplifiers, this principle has to be modified to " maximum undistorted power ", and in such cases the source and the load impedance may be very far from equal.

4.10. FILTER CIRCUITS

Numerous uses are made of the frequency discriminating properties of capacitors and inductors—for example, in the equalisation of Post Office lines and gramophone pick-ups, in the smoothing circuits of mains units, and the tuning of radio receivers. Suppression of low frequencies is achieved by inserting capacitance in series. High frequency attenuation results from inductance in series. The familiar Microphone Correction Unit used to compensate for the

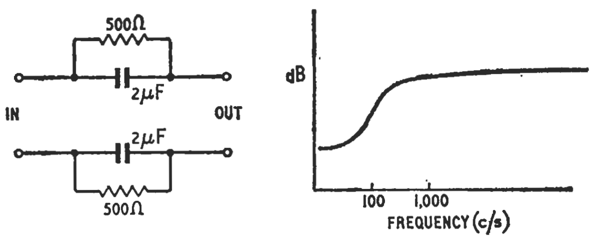


Fig. 4.14. Circuit diagram and response curve of microphone correction unit

rise in bass for close speech on ribbon microphones has a circuit and response graph as shown in Fig. 4.14.

4.11. TUNED CIRCUITS

Since the reactance of an inductor increases with frequency while that of a capacitor decreases, there is one frequency for any combination of the two components at which their reactances are the same.

This is called the *resonant frequency* and the circuit thus formed is called a *tuned circuit*. Resonance in this case is analogous to acoustic resonance mentioned in an earlier chapter, indeed, acoustic calculations can be performed in analogue form using electrical symbols. It is possible to connect L and C either in series, with respect to the generator, or in parallel—see Fig. 4.15. Without going deeply into the theory, it is found that the reactances cancel each other in the series circuit, so that the current rises to a maximum at resonance. In the parallel circuit, the current drawn from the source falls to a minimum at resonance.

The sharpness of resonance depends on the amount of resistance present. If R is high, the circuit is said to be broadly tuned. Pursuing the analogy between electrical and mechanical resonance

(see Chapter 1), resistance in the former corresponds to friction in the latter.

Tuned circuits find application in radio receivers and transmitters, oscillators, and some equaliser circuits.

4.12. DIODE AND TRIODE VALVES

A thermionic valve consists of an evacuated glass or metal tube in which are a number of metal " electrodes " which connect to pins or terminals on the outside. A low voltage, usually about 4 or 6 volts, is connected to two of the pins, so that current will flow through the first of the electrodes, known as the *heater* or *filament*. The heating effect of the current causes electrons to be emitted from a special coating which has been applied either to the filament itself (in directly heated valves) or to a separate electrode called the *cathode* (in indirectly heated valves). Provision of one more

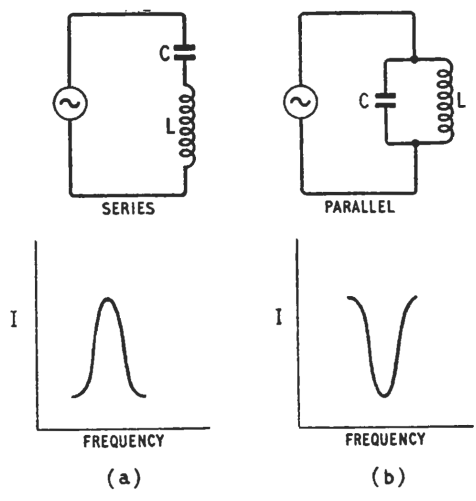


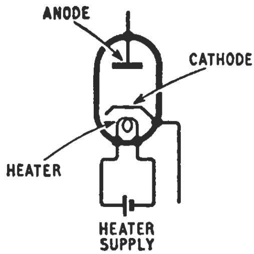
Fig. 4.15. Resonance in (a) series, and (b) in parallel circuits

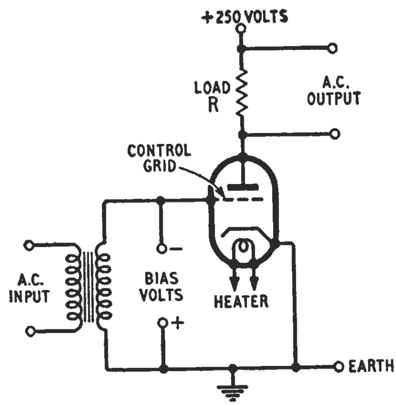
electrode, known as the *anode*, produces the simplest type of valve —a diode (Fig. 4.16).

If an alternating voltage is applied between anode and cathode, pulses of emitted electrons will flow to the anode during its positive half-cycles, but no electrons can flow from anode to cathode in the remaining half-cycles. The diode valve is thus seen to be a one-way device, and one of its principal uses is to convert a.c. into d.c.

Such a rectifier circuit is found in mains units used with condenser microphones.

The *triode* valve is constructed in the same way as the diode, with the addition of a further electrode. This consists of a fine wire





Above: Fig. 4.16. Diode valve.

Right: Fig. 4.17. Triode valve

mesh, called the *control grid*, placed between the cathode and the anode (Fig. 4.17).

The stream of electrons from cathode to anode passes through the grid, which therefore exerts an influence on the current flow. For a given positive voltage on the anode, and " bias " voltage on the grid, a certain value of current will flow through the valve. Because of the grid's closer proximity to the cathode, comparatively small changes in grid voltage will cause large fluctuations in anode current. This is the principle of the *triode amplifier*. Applying an a.c. voltage between grid and cathode swings the anode current up and down in such a way as to produce larger drops in voltage through the load *R*, across which appears an amplified version of the input.

There is a danger in programme circuits of overloading amplifiers. If the swings in grid voltage exceed a certain value, the anode current swings will no longer follow this faithfully (remember that the valve is a one-way device) and serious distortion may result.

Many more complicated types of valve are used in different applications, their names giving a clue to the number of electrodes

they have—for instance, tetrodes, pentodes and hexodes, have 4, 5 and 6 electrodes respectively.

4.13. TRANSISTORS

Since the war a new electronic device which may eventually replace the valve for many applications has been developed to the stage where it is now in use in great quantities. This is the *transistor*, and in several ways it can be said to be similar to the thermionic valve. The common type of transistor is a three-electrode device operating in a very similar manner, at least as far as its external performance is concerned, to a triode.

In its simplest form, the *point-contact transistor* consists of a *base* of a semiconductor material, such as impure germanium, and two point contacts, each like the " cat's whisker " in the early crystal set. These are known respectively as the *emitter* and the *collector*. In more recent forms of the transistor the rather unstable point contacts have been replaced by other pieces of germanium, the effect of the point contact taking place at the junctions between the three pieces. This form, the *junction transistor*, has now almost entirely superseded the point-contact variety and is the type most likely to be found in audio-frequency equipment.

4.13.1. Comparison between the Transistor and the Triode Valve

Both the normal junction transistor and triode valve are three-electrode devices, and in this sense are similar. However, for the valve to operate, the electrodes have to be mounted in an evacuated envelope, usually made of glass, and the vacuum in this envelope has to be maintained as thoroughly as possible. The valve operates by virtue of the flow of electrons from cathode to anode, the rate of this flow being varied by voltages applied to the control grid (see section 4.12). In the case of the transistor no vacuum is necessary, the device functioning at normal atmospheric pressure. The current in this case, instead of flowing through a vacuum is flowing through solid material, and in consequence, the device can be made smaller, stronger and virtually non-microphonic. No heater is necessary as in the valve, and this means that the likelihood of hum induction from a.c. heater supplies is removed. Indeed, troubles of this sort are less likely to occur for various reasons and the device is far less subject to deterioration and failure than the thermionic valve. As mentioned above, in the valve the current is carried by negative electrons; in the transistor the current is carried by positive carriers as well. These positive carriers are referred to as *holes*.

This name arises since they actually are holes in the atomic structure of the material. (They might be said to be similar to the bubble in a spirit level.) The hole is positively charged and is exactly equal in charge to the negative electron. Since the hole and the electron are of opposite sign they will tend to attract each other and the electron may eventually fill the hole. No more current can then flow so far as this particular pair is concerned.

4.14. TYPES OF TRANSISTORS

Transistor material can be produced having any desired number of holes or electrons, the relative proportions of these depending on the impurities present. During the manufacture of the transistor the germanium, or other semiconductor material, is very carefully purified and then, by the addition of minute amounts of selected impurity, the required proportion of holes and electrons can be introduced. When there are more holes than electrons the semi-

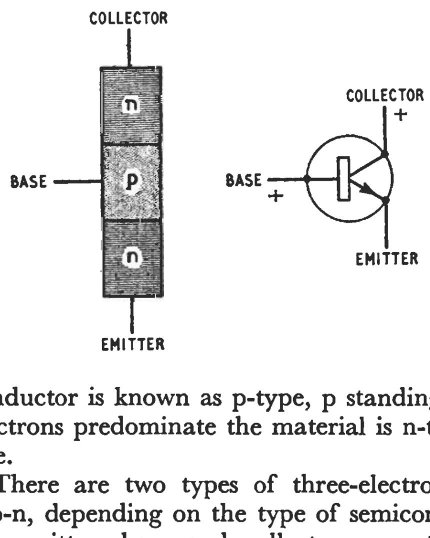


Fig. 4.18. Diagrammatic representation of n–p–n transistor

conductor is known as p-type, p standing for positive. When the electrons predominate the material is n-type, n standing for negative.

There are two types of three-electrode transistor, p-n-p and n-p-n, depending on the type of semiconductor material used for the emitter, base and collector respectively. We will consider first the n-p-n type.

4.14.1. **The N-P-N Transistor** (Fig. 4.18)

In the n-p-n transistor the emitter is made of n-type material and so negative electrons are the predominating carrier. The emitter corresponds roughly to the cathode of a valve, and when the base,

corresponding roughly to the grid, is biased positively with respect to the emitter, electrons can flow from the emitter into the base. If the collector of the transistor is now biased positively to the base the excess of electrons in the base will be absorbed by it. It will be seen that the collector corresponds to the anode of a valve. Now the base is p-type material and so not all the electrons from the emitter will pass on to the collector; some will be absorbed by the base, being combined with its " holes ". If this were to happen indefinitely the base would cease to be p-type material and would not have any charge at all. The charge is maintained by a flow of current from the source of base bias. Varying this current, for example by means of a superimposed programme current, varies the voltage across the base/emitter junction and so varies the current taken by the collector.

4.14.2. P-N-P Transistor (Fig. 4.19)

This is the more common type of transistor, and in this case the emitter and collector are of p-type semi-conductor and the base of n-type. The operation is similar to that described in the n-p-n

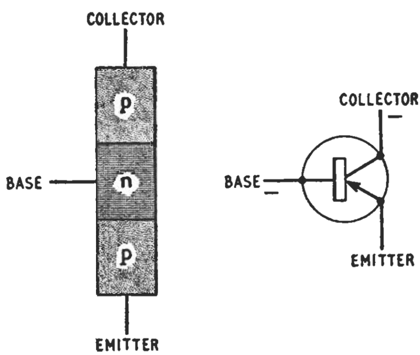


Fig. 4.19. Diagrammatic representation of p–n–p transistor

transistor, but instead of the movement being of electrons, it is a movement of holes, and the base in consequence needs to be biased negatively with respect to the emitter, and the collector negatively in respect to the base.

4.14.3. Practical Circuit

The circuit of a single transistor amplifier is shown in its most elementary form in Fig. 4.20, and along side it the comparative circuit for a triode valve. It will be seen that in many respects

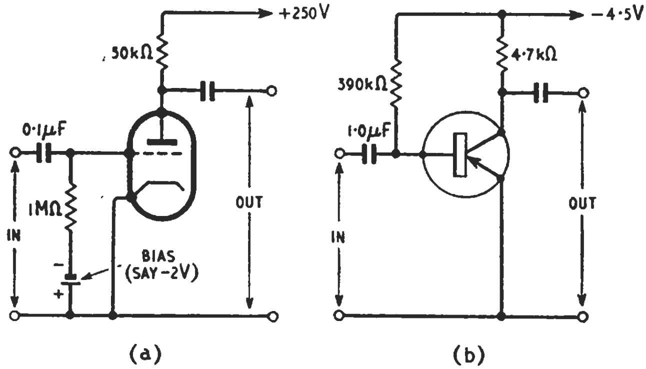


Fig. 4.20. Comparison of transistor and triode circuits

they are similar but it must be noted that in the case of the transistor a variation of input *current* is required to produce an amplified variation in the current through the collector load, whereas in the case of the valve a *voltage* variation at the grid is necessary to produce a current variation in the anode load.

4.15. ADVANTAGES AND DISADVANTAGES OF TRANSISTORS

4.15.1. Advantages

(1) Extremely small size and power consumption making the construction of highly compact equipment possible.

(2) Almost indefinite life under normal conditions.

(3) Freedom from a.c. induced hum, no heater supply being necessary.

(4) Low operating voltages mean that portable equipment can easily be run from dry batteries, and mains operated equipment can be much safer than might be the case with valve equipment.

4.15.2. Disadvantages

(1) The characteristics of the transistor vary considerably with changes in temperature. These variations can fortunately be minimised by suitable circuit design.

(2) In general the level of self-generated noise, usually hiss, in audio-frequency amplifiers using transistors is higher than that in

similar equipment using valves. However, circuit design again can reduce this to an acceptable value.

4.16. TRANSISTORISED EQUIPMENT USED IN THE BBC

Transistorised equipment has so far not come into general use in the BBC, but certain pieces of equipment have been produced in transistorised versions. The E.M.I. midget recorder (see Chapter 10) is now produced in transistorised form, and numbers of these are in service. A series of transistor amplifiers is now being produced and these will be incorporated in new studio equipment. The amplifier AM 9/1 is a transistorised amplifier for outside broadcast use.

5

MICROPHONES

5.1. MICROPHONE REQUIREMENTS FOR BROADCASTING

A microphone is a device for converting sound energy into electrical energy. Amongst the desirable characteristics required of a microphone are:

(1) The conversion should be equally efficient at all frequencies in the working range—i.e., the waveforms of the sound input and electrical output should be substantially the same.

(2) The *output level* should be as high as possible in order that it may overcome unwanted random electrical disturbances known as *noise*, always present in amplifiers.

(3) The *polar characteristic* of the microphone should be the same at all frequencies in the working range.

(4) The *frequency response* of the microphone should be uniform throughout the working range.

(5) Unwanted electrical noise generated by the microphone itself should be low in relation to the wanted signal.

5.2. METHODS OF DERIVING MECHANICAL FORCE

There are two ways of deriving the force which sets the moving parts of a microphone into vibration. They are called *pressure operation* and *pressure-gradient operation.* Sometimes both types of operation will be used in a single microphone.

5.2.1. **Pressure Operation** (Fig. 5.1)

Any microphone whose diaphragm is open to the air on one side only is said to be *pressure operated*, and may be thought of as similar to a quick-acting barometer. The magnitude of the force

on the diaphragm depends on its area and the instantaneous pressure of the sound wave.

If any sort of cavity is formed in front of the diaphragm, the pressure and therefore the frequency characteristic tends to show a peak due to its resonance. Also, owing to the obstacle effect

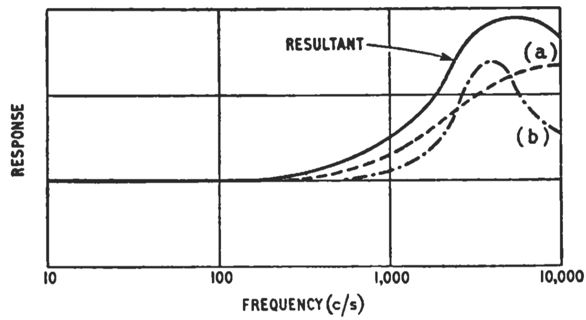


Fig. 5.1. Effect of (a) obstacle effect and (b) resonance of cavity in front of diaphragm on the frequency response of a pressure-operated microphone

(see Chapter 1) an increase tends to result at high frequencies—where the microphone is a reflector of sound waves. This high-frequency rise due to the obstacle effect is used by designers of pressure microphones to compensate for losses in high frequencies which might otherwise occur. In some early pressure microphones

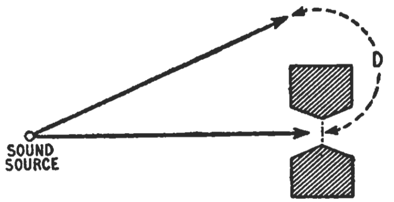


Fig. 5.2. Difference in phase introduced in P.G. operation by extra path length D

it gave rise to a considerable increase in response to high frequencies, and gave these microphones a characteristic " toppiness ".

5.2.2. Pressure-gradient Operation (Fig. 5.2)

When both sides of the diaphragm are open to the air—as in many ribbon microphones—sound waves reach both sides, and the resultant force is due to difference in pressure at the two points.

The magnitude of this force depends on the phase difference between the instantaneous pressures, and this in turn will depend on the path difference *D* and the wavelength. In a given microphone, *D* is a fixed distance (about 1½ inches in a typical microphone), so that the phase difference due to travelling this extra distance is small at long wavelengths and large at short wavelengths.

It follows that the actual force derived in a given P.G. microphone increases with frequency. The designer must arrange that this increasing force nevertheless results in a uniform output.

5.3. BASS TIP-UP IN PRESSURE-GRADIENT OPERATION

When a microphone is placed close to a source of spherical waves, such as the human voice, the inverse square law—whereby the

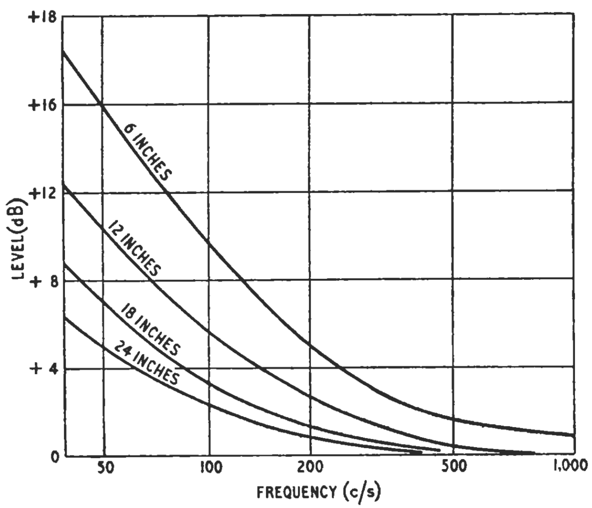


Fig. 5.3. Increase in bass for close working in P.G. operation

sound intensity falls off rapidly with distance—must be taken into account. This has no serious consequence in pressure operation, but in P.G. microphones it means that the pressure difference between front and back due to phase change is augmented by a difference in actual intensity. This is of the greatest significance at low frequencies, since the phase difference is so small. The net effect is summarised in Fig. 5.3—namely, exaggeration of bass frequencies, getting worse as the microphone distance is further

reduced. The minimum distance should be about 2 ft, except for special effects.

5.4. DIRECTIONAL PROPERTIES

The *polar diagram* of a microphone (or loudspeaker) is a graph of the relative voltage output for sounds arriving at different angles. It is usual to measure angles from the microphone axis, and to draw a separate polar diagram for various frequencies.

The three basic diagrams met in practice are:—

(a) circle—omni-directional (associated with pressure operation);

(b) figure-of-eight—two-sided (associated with pressure-gradient operation);

(c) cardioid—single-sided (associated with pressure and pressure-gradient operation combined).

Intermediate characteristics can also be obtained by combining pressure and pressure-gradient operation in different proportions.

5.4.1. Pressure Microphones

All pressure-operated microphones have a similar set of polar diagrams—omni-directional at low frequencies, and tending more to one-sided response at high frequencies. Such differences as

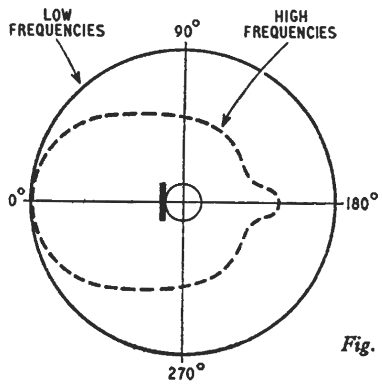


Fig. 5.4. Polar diagram for pressure-operated microphones

occur are due to the size and shape of the microphone, and not to the particular method of generating the voltage.

When the microphone is small compared with the wavelength—sounds up to about 1,000 c/s—no reflection takes place, sounds

arrive at the diaphragm equally from all angles, and the polar diagram is a circle (actually a sphere, if three dimensions are considered).

At higher frequencies, the pattern changes, due to a combination of two effects, firstly the *obstacle effect* mentioned earlier, and secondly the fact that at oblique angles a sound will arrive at different parts of the diaphragm at different times. When the frequency is high, and the wavelength shorter than the diameter of the diaphragm, these arrivals may be out of phase, causing partial cancellation.

The overall result for a microphone about three inches in diameter is to give an acceptance angle with good frequency response of about 60° and progressive loss of high frequencies at angles greater than this (Fig. 5.4).

The total indirect sound picked up by a pressure operated microphone tends for the above reasons to contain less top than middle and bass. The result of this is that in reverberant conditions the sound produced appears to be lacking in high frequencies, unless a close technique can be used.

Modern pressure operated microphones are exceedingly small, some having diaphragm diameters as little as half an inch, and in these the effect is quite small. It can be further minimised by fitting a small baffle plate or "biscuit" to reduce the high frequency response on the front axis.

5.4.2. Pressure-Gradient Microphones

The force in a pressure-gradient microphone due to phase difference at front and back has been shown to depend on the

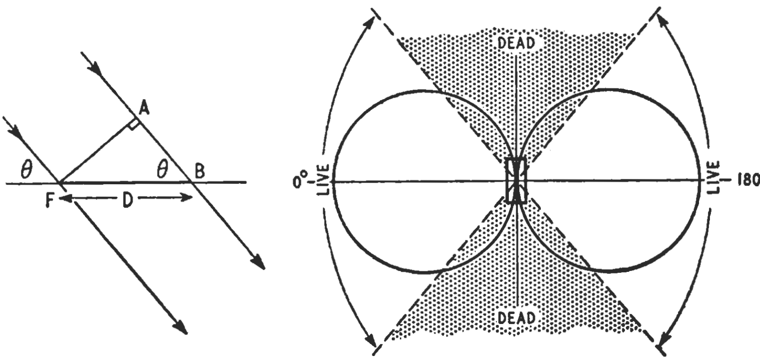


Fig. 5.5. Polar diagram for pressure-gradient microphones, showing (a) reduction in effective path length at oblique angles and (b) two 100° "live" angles, and two 80° "dead" angles

extra path length. This in turn depends on the angle of incidence, as is shown by the diagram, being a maximum at 0° and 180°, and falling to zero at 90° and 270°. Thus the microphone is equally sensitive on both faces but is " dead" at the sides (Fig. 5.5).

For an angle of incidence θ the extra path length is not $FB(=D)$, but AB which becomes smaller as the angle increases. Mathematically-inclined readers may notice that

$$\frac{AB}{FB} = \cos\theta$$

$$\therefore \quad AB = D\cos\theta$$

and the force, and therefore the output, will be proportional to the cosine of the angle of incidence.

By symmetry in the other three quadrants, we arrive at the familiar figure-of-eight diagram.

5.4.3. Cardioid Microphones

A heart-shaped (cardioid) polar diagram has become popular where one-sided pick-up over a wide angle is required—for discussions, choirs, sections of an orchestra etc. It may be obtained

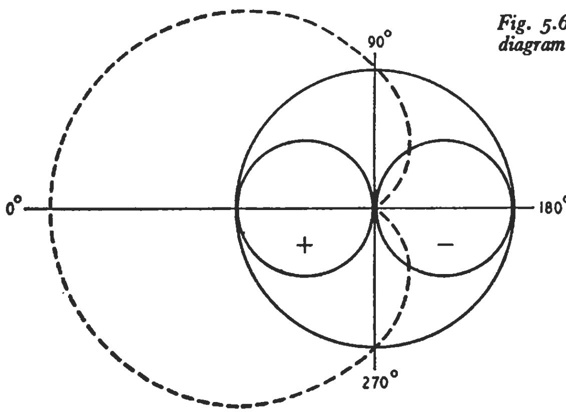


Fig. 5.6. Derivation of cardioid diagram by addition of circle and figure-of-eight

by combining the characteristics of an omni-directional element and a pressure-gradient element whose maximum outputs are equal.

This is shown diagrammatically in Fig. 5.6. The two elements are taken to be *in phase* to the left of the diagram, so that their outputs reinforce each other. The phase reversal which occurs on sounds arriving at the back of the figure-of-eight element causes

the outputs of the two elements to be in opposition, so that complete cancellation takes place at 180°, and the combined output falls to zero. Unfortunately, the polar diagram of the cardioid microphone which is constructed from two separate elements will tend

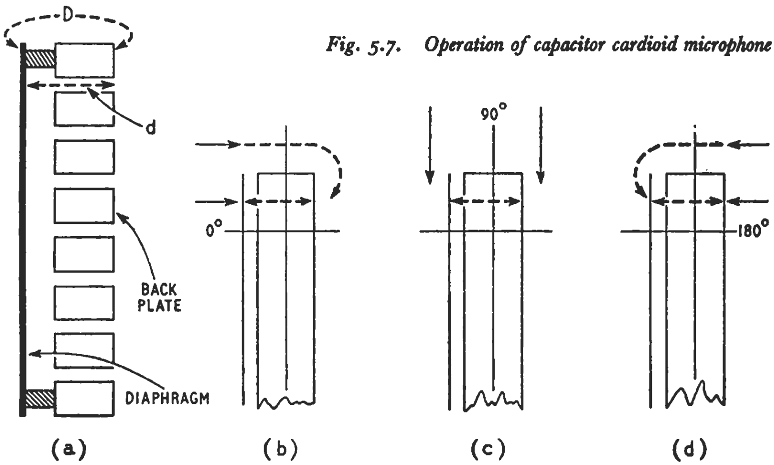


Fig. 5.7. Operation of capacitor cardioid microphone

to vary with frequency. This is partly due to the difficulty in constructing a pressure operated element which is omni-directional at all frequencies. Also, it is necessary for the two elements to possess identical response curves, which is difficult to arrange.

A better maintained cardioid response is possible when a single element can be made to combine the characteristics of pressure and pressure-gradient operation. Electrostatic microphones having cardioid response working on this principle have been used in broadcasting since about 1935. Cardioid ribbon microphones, which operate by partially enclosing the rear of the ribbon to give a combination of pressure and pressure-gradient characteristic, have been available since the war. More recently, since about 1955, the principle has been applied to moving-coil microphones, although, as in the case of electrostatic microphones it was not enclosing that was necessary but opening a path to the rear of the diaphragm.

A general description of the process will be given by reference to Fig. 5.7. In this the electrostatic cardioid microphone capsule is described, but the same principles apply to the other versions.

Pressure-gradient operation is introduced by boring a pattern of holes through the back plate of the microphone, so that sounds may

act on the back as well as the front of the diaphragm. The extra path length in this case (on which the force on the diaphragm will depend) is made up of two parts:—

D—measured on the outside—which will vary according to the direction of the sound source (c.f. pressure-gradient operation) and

d—measured through the back plate—which is independent of the direction of the sound source (c.f. pressure operation).

The overall result is a cardioid-shaped polar diagram, since the extra path travelled by sounds to reach the back of the diaphragm (compared with the front) is $(D + d)d$, and O respectively, for sounds at angles 0°, 90°, and 180°. See Fig. 5.7 (b), (c), and (d).

The actual length of *d* is likely to be rather less than *D*, but by careful design of the back plate its effective length can be increased until *d* and *D* are equal, so that complete cancellation of sounds from 180° is achieved. This increase in the effective length of *d* can be obtained by making the back plate from two discs with the pattern of holes not quite opposite each other. Rotating one disc in relation to the other during the manufacturing process allows the necessary adjustment to be made.

Since a cardioid microphone can be said to be a combination of pressure and pressure-gradient microphones, there will be a slight bass rise under close working conditions, due to the pressure-gradient part of the device. This will not be so serious as that which occurs when pressure-gradient operation is used alone.

5.4.4. Variable Polar Diagram Microphones

By using two of the cardioid microphone capsules described above, mounted back to back, it is possible to produce a number of polar responses from the combined pair. This is achieved by the addition of the output of the two capsules in differing amounts either in or out of phase. The polar characteristics available by this means can vary from omni-directional through cardioid to figure-of-eight.

Electrostatic microphones of this type are in practice constructed with two diaphragms, one on either side of a common back plate (see A.K.G. microphone type C.12).

5.5. METHODS OF DERIVING THE VOLTAGE

There are at least five methods of deriving an electrical output from the vibrations of the diaphragm. The electrical action in each of these cases will be dealt with briefly before giving descriptions of microphones which are in current use.

5.5.1. Carbon Microphone

Variations in pressure in the sound waves cause the diaphragm to vibrate. The resultant alternating pressure on the carbon granules compresses and releases them, causing the resistance between the terminals to alternate about its mean value. This

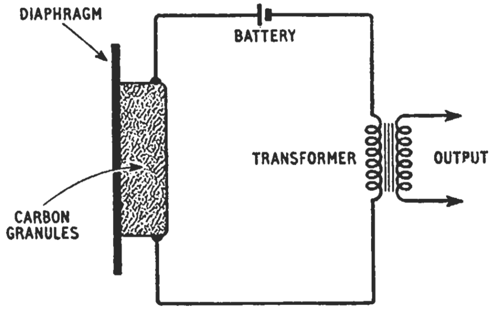


Fig. 5.8. The carbon microphone

imposes an alternating component on the steady current drawn from the battery. Using a transformer, the a.c. component may be tapped off as output (Fig. 5.8).

Advantages of the carbon microphone are its robust construction and large output. These make it suitable for use in the mouthpiece of telephones. Disadvantages are that a battery supply is needed, that it generates a high amount of electrical noise, that the granules tend to " pack " or stick together, and that the output volume and directional pattern tend to vary with frequency.

5.5.2. Moving-coil Microphone

This microphone, sometimes called the dynamic microphone, works on the electromagnetic principle, and was briefly discussed in Chapter 4. As the diaphragm vibrates in sympathy with the sound waves, the coil which is fixed to it is made to vibrate in the field of a strong permanent magnet. The resultant induced voltage will be at the frequency of the sound, and its strength will vary in accordance with that of the sound waves. The ends of the coil are usually connected to a pair of insulated terminals. The impedance of the coil is low—usually of the order of 20 or 30 ohms, so that a " matching " transformer is necessary to step it up to 300 ohms when used with BBC amplifiers (Fig. 5.9).

Moving-coil microphones are robust, need no special amplifiers, and can give good quality. They are therefore suitable for outside

broadcasts and as hand microphones. Most moving-coil microphones are pressure operated, and so are nominally omni-directional. However, some are rather large in size and variation in polar diagram may set in as low as 1,000 c/s. Moving-coil microphones

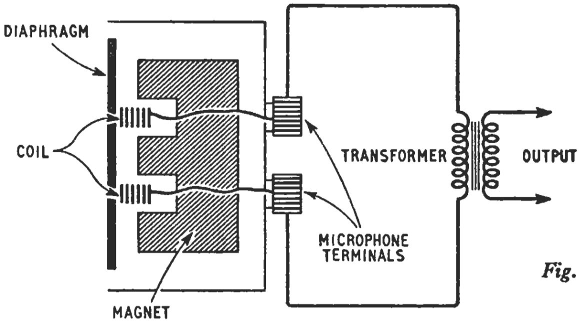


Fig. 5.9. The moving-coil microphone

are also produced having a cardioid polar diagram as previously described.

5.5.3. Ribbon Microphone

This microphone also works on the electromagnetic principle. A ribbon of metal foil is suspended between special pole-pieces attached to a strong permanent magnet, and combines the functions of diaphragm and moving conductor. The ribbon is corrugated, and loosely tensioned between insulated clamps. Nearly all ribbon

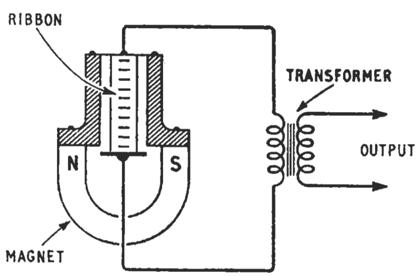


Fig. 5.10. The ribbon microphone

microphones in the BBC are of the pressure-gradient type, being open to the air on both sides, and having a figure-of-eight polar diagram. Ribbon microphones can, however, be produced having pressure characteristics and also a combination of pressure and

pressure gradient, so that a number of polar responses can be obtained from this type of operation. The impedance of the ribbon is low—a fraction of 1 ohm—so that a matching transformer has to be built into the microphone to step up the impedance to 300 ohms (Fig. 5.10).

Advantages of the ribbon microphone are the excellent quality that is possible, and the fact that the figure-of-eight polar diagram is maintained at all frequencies. It is therefore useful as a general purpose studio microphone—for talks, drama and music.

Its disadvantages are that it is very susceptible to wind noise and vibration, and cannot be used close to a sound source because of the inherent bass rise under these conditions.

5.5.4. Electrostatic Microphone (Condenser Microphone or Capacitor Microphone)*

This microphone is really a capacitor, with one plate fixed and the other a flexible diaphragm. As the diaphragm vibrates, the capacitance varies about its mean value. The resistance R (see

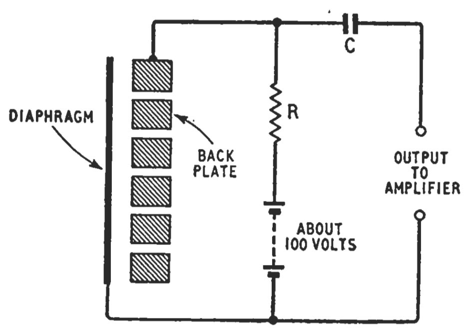


Fig. 5.11. The electrostatic microphone

Fig. 5.11) is high in value so that the charge on the microphone may be regarded as constant. Under these conditions an alternating voltage is produced proportional to the fluctuating capacitance.

The microphone capacitance is very small, and corresponds to a very high impedance. It is therefore necessary to mount an amplifier at the microphone head, since even a short length of lead might cause a serious loss in output. The amplifier is not primarily used to step up the voltage, but to match the higher impedance of

* A condenser is now called a capacitor in electronic practice; the term " condenser microphone " is therefore inconsistent and is being replaced by " electrostatic microphone " or " capacitor microphone ".

the microphone to the 300 ohms impedance of the studio circuit. The coupling capacitor *C* isolates the amplifier from the battery (about 100 volts). For good sensitivity, the spacing between diaphragm and back plate is made very small, and the latter is slotted. A vent hole is provided to equalise slow changes in pressure—similar to the way in which the eustachian tube in the human ear opens periodically to equalise inside and outside pressures.

Advantages of the electrostatic microphone are that high quality is possible and that in some modern types a series of useful directional patterns may be achieved in a single unit. A disadvantage is that a head amplifier is necessary, with its associated power supplies and multi-cored cables.

5.5.5. Crystal Microphone

Some materials, when subjected to mechanical strain, produce an electrical voltage across opposite faces. This is known as the *piezo-electric* effect and is made use of in the crystal microphone. A sandwich made of two slices from a crystal of Rochelle salt is arranged so that the sound causes it to bend or twist, thereby producing a voltage corresponding to the sound input. The crystal movement

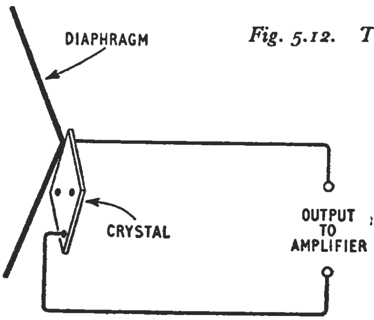


Fig. 5.12. The crystal microphone

can either be accomplished by means of a diaphragm attached to the crystal, or by the sound impinging on the crystal itself. In the drawing a diaphragm is attached to one corner of a crystal assembly, while the other three corners are fixed (Fig. 5.12).

Advantages of the crystal microphone are that high output is possible, and the bulk can be reduced to give omni-directional characteristics at all frequencies. A disadvantage is that the high internal impedance makes it necessary to use low capacity cable to connect the microphone to the grid of the amplifier valve, if

high frequency losses are to be avoided. This limits the length of cable that can be used. All crystal microphones normally met are pressure operated, although some pressure-gradient types are available.

5.6. MICROPHONES IN CURRENT USE

The last few years have seen a great deal of activity in the field of microphone design. The introduction of new types to BBC studios is necessarily gradual, and old and new types are therefore likely to be used side by side for some time to come. The descriptive notes which follow are intended to help in identifying the different types and in choosing the best from those available for different applications. The order of treatment will be the same as in the previous section. Advice on the positioning of microphones is contained in Chapter 13.

5.6.1. Carbon Microphones

No carbon microphones are used in BBC studios.

5.6.2. Moving-coil Microphones

A series of microphones of this type has been manufactured by Standard Telephones and Cables Limited, and will be given the S.T. & C. type numbers for easy reference (Plate 5.1).

(a) S.T. & C. 4017C (obsolete)

On account of its rugged construction, this microphone has been widely used on a hand-grip for interviews, etc.

It is common practice to speak across rather than directly into the diaphragm to avoid excessive high frequency response. The terminals require careful tightening as the cable tends to work loose.

The impedance of the microphone is 25 ohms and so a separate matching transformer is necessary in 300 ohm circuits, and is especially important if the 4017 is to be " mixed " with other types. The output level is approximately 4 dB lower than that of the 4038 ribbon microphone. The weight is 2½ lb. This microphone is now regarded as obsolete, the replacement types being the S.T. & C. 4032 and 4035, which weigh about ¾ lb.

(b) S.T. & C. 4021A and 4021F (" Apple and Biscuit ")

This microphone was designed to give omni-directional characteristics at high as well as low frequencies. The case is therefore

spherical, and reduced in size. The " biscuit " does much towards maintaining all-round response; high frequency sounds from the top are attenuated and those from underneath are partially reflected on to the diaphragm. This counteracts the high frequency rise due to the obstacle effect mentioned earlier.

There are few studio applications where an omni-directional microphone is used at present, since it is more difficult to achieve a satisfactory ratio of direct to indirect sound under reverberant acoustic conditions than with a figure-of-eight or cardioid microphone. The 4021 finds its use in outdoor applications, in echo rooms and as the talk-back microphone in studios.

(*c*) *S.T. & C. 4032C and 4032D*

This is designed to replace the 4017 for hand use only, is smaller and lighter, and is mounted in a streamlined hand-grip. It has an improved frequency response, but has the same general directional properties. Its internal construction is similar to that of the 4021. It weighs ¾ lb. A windshield is available which will fit either the 4032 or the 4035.

The 4032D is identical with the above, except that a transformer fitted inside the handle provides an output impedance of 7,000 ohms for direct connection to the E.M.I. Midget Recorder.

(*d*) *S.T. & C. 4035A*

This is the general purpose replacement for the obsolete 4017C microphone. It, too, is smaller and lighter and has the same directional properties with an improved frequency response. It is useful for outdoor work of all kinds, and weighs 1 lb including the cable-connecting jack.

(*e*) *S.T. & C. 4037A*

The diaphragm and magnet system of this microphone is similar to the 4021 or " apple and biscuit ". It is fitted in a " stick " tubular mounting about 1 in. in diameter and 8 in. long. The frequency response may suffer if the vent holes at the base of the microphone are covered, although ordinary handling will not usually cause trouble.

Two types of windshield are provided, a small plastic one made by Standard Telephones and Cables Limited and a larger one, BBC designed, of wire mesh. It has been found that further improvement under windy conditions is obtained if a " bung " is fitted



Plate 5.1. S.T. & C. moving coil microphones—type 4037, 4021, 4035 and 4032.



Plate 5.2. Ribbon microphones—type AXBT and PGS



Plate 5.3. S.T. & C. microphone—type 4038.

Below: Plate 5.4. S.T. & C. type 4104—lip ribbon microphone





Plate 5·5. Reslo ribbon microphone type VRM T

Below: Plate 5.6. S.T. & C. 4033A cardioid microphone





Above and left: Plate 5.7. Neumann electrostatic microphone type KM54a and close-speaking windshield

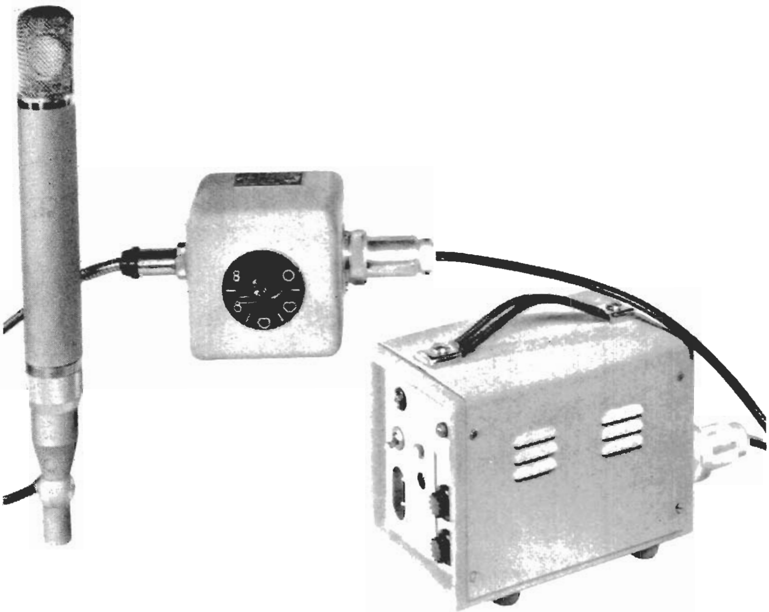


Plate 5.8. C. 12 microphone, polar diagram selector and supply unit

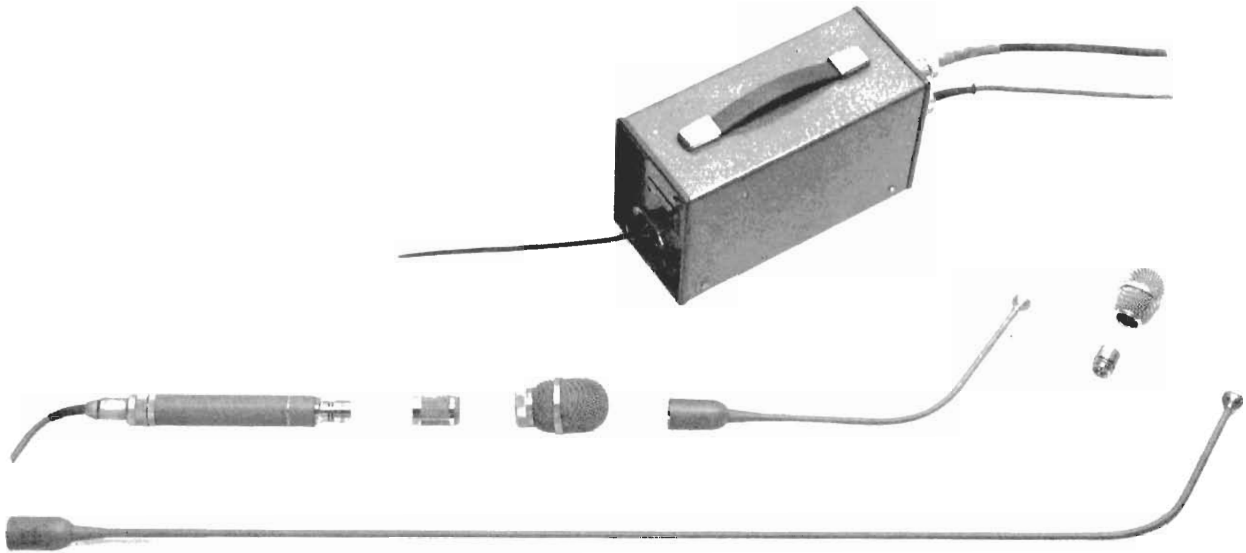


Plate 5.9. A.K.G.C. 28 microphone and mains unit and A.K.G.C. 29, C. 30 extension pieces



Plate 5.10. K.M. 56 microphone

inside the microphone to restrict the passage of wind through the vent holes. The slight bass loss which is introduced by the bung is not normally serious for speech.

The 4037A, and more particularly the shorter but otherwise similar 4037B is inconspicuous in use, and these microphones therefore have recommended themselves to television interviewers and dance band vocalists. The 4037A weighs approximately ¾ lb with cable jack, and the 4037B just over ½ lb.

(f) R.C.A. BK6B

This is a small moving-coil microphone for " personal " use, and is intended to be hung round the neck of the user, a lanyard being provided for this purpose. The microphone can be conveniently positioned beneath a man's tie, and is then reasonably inconspicuous for television use. The frequency characteristics of the microphone have been designed to compensate for the somewhat unusual position of the microphone in relation to the mouth of the speaker and for the fact that it may be covered by clothing.

With all personal microphones, some such correction is generally necessary. Moreover, if the microphone has to be used in a programme where the speaker is using other static microphones, special equalisation, depending on the studio conditions and the person concerned, may be desirable in order to prevent too great a change in speech quality when the two types are used consecutively.

5.6.3. Ribbon Microphones

There are seven versions of the ribbon microphone in current use—the older types AXB and AXBT, the P.G.S., S.T. & C. Type 4038, Reslo VRM/T and finally the BBC Lip Ribbons L1 and L2 (Plates 5.2, 3 and 4).

(a) BBC-Marconi Type AXB and AXBT

The AXB microphone, which is a BBC design, came into service in 1934, and the later version, AXBT, in 1943. These are identical in appearance, except for a white " T " painted on the AXBT. The latter has a stronger magnet, and in consequence has 6 dB greater output, and musical " attack " appears to be better reproduced. The AXB type is now being withdrawn.

In the vertical plane, the acceptance angle is much narrower for high frequencies than for low frequencies and is not symmetrical, so that different quality is obtained above and below the axis. The

angle of tilt is therefore very important, and angles in excess of about 20° to the source result in a loss of top.

For discussion programmes and for use in theatres etc., the type AXBT microphone is often felt to be too large and conspicuous.

(*b*) *P.G.S.* (*Pressure-gradient Single*) *

New magnetic materials have made possible the design of smaller and lighter microphones, and the BBC Research Department has produced the P.G.S. ribbon microphone, which is less than one-third the weight of the AXBT (2·5 lb, compared with 9·2 lb). Its output level is 4 dB less than the AXBT, but it has a much improved frequency response—the bass is better maintained, and the top extends about half an octave higher. To keep the size down to a minimum, the case fits closely around the magnet and the matching transformer is inside the base of the stirrup. The ribbon is 1 in. long, compared with 2½ in. in the AXBT.

The variation in high frequency response with angle of tilt, noticed in the Type AXBT microphone, is less in the P.G.S. The production models of the P.G.S. are manufactured by Standard Telephones and Cables Limited. Three type numbers are in existence but only the last named will normally be met in BBC sound studios:—

4038A—Black case,	30	ohms output impedance
4038B—Bronze case,	30	,, ,, ,,
4038C—Bronze case,	300	,, ,, ,,

In BBC television studios the 4038B is in regular use since many of the microphone circuits are of 30 ohms impedance, as opposed to 300 ohms in sound service. To help identify the microphones, even in badly-lit conditions, the 4038B has a small recess engraved "30 ohms", and the 4038C has a small raised button engraved "300 ohms".

(*c*) *Reslo VRM/T*

This is a miniature pressure-gradient ribbon microphone having a nominal figure-of-eight response. It is very small in size and therefore has become popular for use in television studios, where

* "Single" refers to the fact that this microphone has a single horseshoe magnet. The term was used to differentiate between this microphone and an experimental type known as the P.G.D. (D for double) which had two magnets. The P.G.D. was not produced commercially.

it can be used as an inconspicuous " inshot " microphone. It is a modified version of the outwardly similar Reslo RBM/T.

(*d*) *Noise-cancelling Lip-Ribbon Microphones* (*L*1—1937 *design; L*2—1951 *design*)

For sports commentaries etc., a very close speaking distance is necessary to discriminate against background noise. In the BBC-Marconi lip-ribbon microphone L1, a guard ring gives a speaking distance of 2½ in., and the resultant rise in bass response is corrected by means of acoustic impedances (supplemented by electrical filters as described below). This bass cut attenuates a large part of the crowd noise and gives better speech-to-noise separation.

In the L1, a cradle of eight springs gives protection against bumps, and the base of the horseshoe magnet protects the ribbon from the direct draughts from the speaker's mouth. The carrying-case is specially made to clamp the springs securely, and the microphone should always be transported in this way.

A three-position switch in the carrying-case gives conditions as follows:—

Position No. 3—unequalised, Positions 2 and 1—different degrees of bass cut; these positions are intended for loud, average, and quiet speech. Quiet speech requires the most bass cut, since the frequency content of speech under these conditions is bassy anyway.

An alternative noise-cancelling microphone known as the L2 is now in service. It is of BBC design and manufacture, and has an improved frequency response, and 5 dB greater output. It weighs 1 lb, compared with 1·8 lb for the earlier model. The hand-grip is oval in cross-section, and the angle it makes with the microphone may be varied to suit the individual commentator. The cradle of springs was found to be unnecessary in this model, provided reasonable care is taken in handling the live microphone.

Production models of this lip microphone are manufactured by Standard Telephones and Cables Limited and the type number is 4104A.

A new version of this microphone has now been produced by S.T. & C. It is designated 4104B and has a slightly wider frequency response than the earlier models. It does not have a separate bass equaliser, equalisation similar to the " average "

position being built into the microphone. The 4104B can be recognised by the fact that it has only a single bar as a mouth guard. A third version, 4104C, is similar but of 300 ohms impedance.

5.6.4. S.T. & C. 4033A (Cardioid) (Plate 5.6)

This microphone combines a ribbon element and a small moving-coil element of the 4021 type. A screwdriver-operated switch gives the following conditions:—

P—(Pressure)—omni-directional, using moving-coil only.

R—(Ribbon)—two-sided, using ribbon only.

C—(Cardioid)—single-sided, combining both elements.

As was stated earlier, constructional difficulties are likely when ribbon and moving-coil units are combined in this way. Practical mechanical difficulties prevent the moving-coil unit from being omni-directional at all frequencies. A compromise is necessary, and in this case the outputs of the moving-coil and ribbon units are combined in a suitable electrical network in such a way that as the frequency rises the effect of the ribbon unit is attenuated. By this means the moving-coil element contributes the whole of the output at high frequencies and a good compromise cardioid response is obtained.

The rugged nature of this microphone makes it especially suitable as a boom microphone in television. The ribbon is of unusual stiffness to reduce mechanical bumps, and a shockproof mounting is fitted. A matching transformer is required in 300 ohm circuits, and gives an output level similar to the 4038. In the cardioid position there is a small increase in bass, as would be expected with close working, but the overall frequency characteristic is such that the increase is unimportant at distances over 9 in.

5.6.5. Electrostatic Microphones

All electrostatic microphones need an amplifier close to the diaphragm. In some designs it has been possible to move it a short distance away, up to 3 ft, by the use of special very low capacity cable. The need for an amplifier complicates and enlarges the microphone, involves high and low voltage supplies, and a multi-core connecting cable. In recent types miniature valves and components have made possible reduced size, and mains units have replaced the batteries. The types of microphone which will

be described are the Neumann KM54a and KM56, and the A.K.G. C.12, C.28, C.29 and C.30.

(*a*) *Neumann KM54a* (Plate 5.7)

The KM54a consists of a capsule having a metal diaphragm mounted at the end of a small cylindrical case containing the head amplifier. The axis of the microphone is end-on, and the directional properties are nominally cardioid, but the high frequency response is found to change somewhat at oblique angles. The rejection of unwanted sounds from the back is a very desirable feature in multi-microphone layouts.

A windshield may be necessary, and two types are available. The makers' windshield, described by them as a " close talking shield ", modifies the response and polar diagram, introducing some bass cut, which may be an advantage in the case of " crooners ". The alternative type, a BBC design, does not impair the frequency response, and can safely be used when close working is not required, without affecting the quality. When fitting either windshield care should be taken to see that it is pushed gently on to the microphone as far as it will go.

(*b*) *A.K.G. C.12*

This microphone is a product of the Akustische und Kinogeräte G.m.b.H. of Vienna. It is an electrostatic microphone carrying effectively two cardioid capsules with a common back plate, this combined capsule being spring mounted in a long narrow case containing a head amplifier. The microphone axis is side-on to the amplifier case.

The head amplifier/microphone unit is connected by 65 ft of cable to a mains unit supplying polarising voltage for the microphone capsule and operating voltages for the amplifier. A further cable leads from this mains unit to a polar diagram selector unit and the output of the microphone is taken from a socket on this unit (Plate 5.8).

The polar diagram selection unit allows remote selection of the directional pattern of the microphone from omni-directional through cardioid to figure-of-eight, with three intermediate conditions between each—nine switch positions in all. The operation of this polar diagram selector will be described with reference to Fig. 5.13. It will be seen that the fixed back plate of the two capsules is maintained at a steady potential, exactly half of that of the

available polarising voltage. If the available voltage is 100 volts * this fixed plate is thus maintained at 50 volts. The front diaphragm of the microphone is at zero potential, that is, it is earthed, and so a constant 50 volts difference is maintained on the front half of the microphone. The rear diaphragm of the microphone is connected to the slider of a linear potentiometer which is connected across the whole of the available polarising voltage so that this diaphragm can have any voltage between zero and one

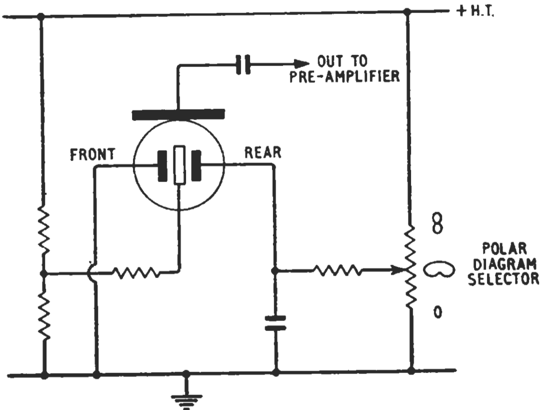


Fig. 5.13. Simplified circuit diagram of A.K.G. C.12 microphone

hundred applied to it. It will thus be seen that the diaphragm can be either 50 volts positive, zero, or 50 volts negative, with respect to the centre plate. Obviously, intermediate positions are also possible.

(*c*) *A.K.G. C.*28 (Plate 5.9)

This microphone is a single diaphragm electrostatic microphone similar in operation to the KM54. The microphone/head amplifier unit is rather larger than the KM54, but not so large as the C.12. The polar response is cardioid, with the microphone axis " end on " to the case.

The microphone has a good bass response and a smooth high frequency response. A windshield, W.28, is available, but even without it the C.28 is not seriously prone to " popping " in close use.

* The actual voltage is 105, but round figures have been used to simplify the explanation.

(*d*) *A.K.G. C.26*

This is similar in appearance to the C.28, but has a capsule which is omni-directional in characteristic.

(*e*) *A.K.G. C.29 and C.30* (Plate 5.9)

These are similar in performance to the C.28, but an extension piece has been fitted between the microphone capsule and the head amplifier unit. In the C.29 this extension is 12 in. long, and in the C.30 it is 36 in. long. These microphones are useful as stage microphones where an unobstrusive instrument is desirable so as not to block the vision of an artist from the audience. A windshield is available for both.

A possible source of confusion arises in that microphones of the outward appearance of the C.29 and C.30 can in fact accommodate the omni-directional capsule normally fitted to the C.26; in fact all the parts of these microphones are physically interchangeable. The different types can be recognised with certainty only by observing the number engraved on the capsule after removing the windshield.

(*f*) *Neumann KM56* (Plate 5.10)

This is similar in appearance to the KM54a microphone, but has a double cardioid capsule similar to that of the C.12, and the axis of the microphone is " side-on ". This enables it to have variable polar diagram characteristics, and the control for the polar diagram is found on the microphone itself. Unlike the C.12 no intermediate switch positions are available.

5.6.6. Crystal Microphones

Crystal microphones are not met a great deal in broadcasting work, but are often found with domestic tape recorders. Some types, however, are used by the BBC, and these will be described.

(*a*) *Acos Lapel* 28/1

This microphone is used in Mobile and O.B. work, where a commentator must be able to move about and have both hands free. It has a rubber-covered case about 1½ in. square, which pins on to the clothing. There is good insulation against mechanical shock, but the cable itself should be pinned to the wearer's clothing a few inches from the microphone to prevent it generating noise. A short length of cable connects the microphone to the pocket pre-amplifier. The power supplies for this pre-amplifier are provided

by a 15 volt battery, and a single pen cell. There is an ON/OFF switch, and a total of 5 hours' continuous operation is possible. As the batteries are cheap, it is recommended that a fresh set be fitted for each programme commitment, discarded batteries being used up on rehearsals.

(*b*) *Acos Mic.*39

This is a small "stick" type microphone, with the crystal capsule mounted at the end of a plastic tubular case about 5 in. long. It is used in the BBC with the Self-Operated Outside Broadcast Equipment.

(*c*) *Ronette MC*65

This is a crystal microphone capsule which for BBC use has been mounted in a plastic case of the "stick" type about 5 in. long. The microphone is of high impedance and has been used by the BBC with miniature tape recorders.

(*d*) *Double Lapel Microphone MC*/4

This consists of two Ronette crystal capsules each mounted in a small metal case with a clip, intended to be fixed to the lapels of a man's jacket, one on each side. The capsules are connected in parallel, and this arrangement helps to prevent loss of volume as the speaker's head is turned from side to side. The output impedance is again high, and the use of the miniature pocket pre-amplifier necessary.

6

LOUDSPEAKERS

A MICROPHONE converts sound energy into electrical energy, and this output " follows " the sound in magnitude and rate of vibration. A loudspeaker reverses this process, being driven by the electrical energy (suitably amplified), and made to vibrate in the same manner as the original source of sound.

The design of a loudspeaker to radiate a wide range of frequencies presents a number of technical problems, and some of these are briefly discussed, followed by short descriptions of BBC loudspeakers in current use.

6.1. MOVING-COIL LOUDSPEAKERS

6.1.1. General

The moving-coil loudspeaker employs a strong magnet, specially shaped to concentrate the magnetic field in a narrow, ring-shaped

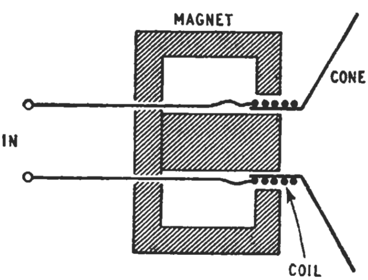


Fig. 6.1. Moving-coil loudspeaker

gap. The speech coil is suspended in this gap and fastened to a conical diaphragm which is usually moulded from paper pulp (Fig. 6.1).

When programme currents are passed through the coil, interaction with the field of the magnet sets the coil and cone into

vibration. The cone will then radiate sound waves at the frequencies present in the current.

How closely this radiation reproduces the original sound is, of course, a matter of careful design.

6.1.2. Directional Effects

The first complication in achieving the ideal loudspeaker is with regard to the directional pattern. As was seen in Chapter 1, a radiator of waves will propagate in all directions or in a relatively

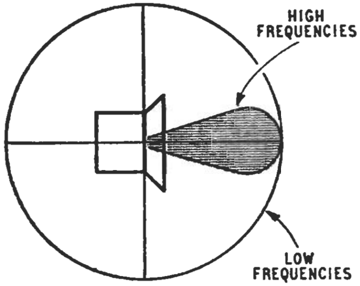


Fig. 6.2. Directional pattern of loudspeaker

narrow beam, according to whether it is small or large compared with the radiated wavelength. But a normal loudspeaker is intermediate in size in relation to the range of sound wavelengths (for 50 c/s and 10,000 c/s the wavelength in air is about 22 ft. and 1½ in. respectively). This means that low frequency sounds are sent out in all directions, while higher frequencies tend to be confined to a narrow angle about the axis of the loudspeaker (Fig. 6.2).

The effect on listening is that best results are obtained directly in front of the loudspeaker, and listening at the side gives the effect of " top cut ". Furthermore, reverberant sound due to the existing room acoustic will have very little " top ".

6.1.3. The Baffle or Enclosure

Another factor which limits the efficiency of a loudspeaker, particularly at low frequencies, is the interference between the back and front radiations. Referring to Fig. 6.1 we see that a forward movement of the cone will cause a compression of the air in front, and an expansion at the back. The net result is the simultaneous radiation of two waves in anti-phase, which will cancel each other

unless the path from back to front is made greater than half-a-wavelength for the lowest frequency required.

A 2 ft 6 in.-square *baffle* (see LSU/4A, Section 6.3.1.) will prevent this cancellation for frequencies down to about 200 c/s (wavelength 5 ft), but short of building the loudspeaker into a wall, a baffle provides only a limited solution.

The addition of sides to form a *box baffle* can give improved bass efficiency, provided steps are taken to reduce the boominess due to resonance of the enclosed air—for example, by the insertion of a lining of glass wool. The back of such baffles is left open to reduce resonance. A *vented enclosure*, however (see LSU/10, Section 6.3.2.) has a closed back, and an aperture by means of which the low frequency resonances can be controlled (Fig. 6.3).

6.1.4. Multi-unit Loudspeakers

For efficient radiation of low frequencies the cone should be large, but the time taken for the coil vibrations to travel out to the edge of a large cone may correspond to one or more cycles at high frequencies. This means that parts of the cone are radiating in anti-phase, with consequent loss of efficiency.

It is for this reason that all wide range monitoring loudspeakers—e.g. LSU/10—incorporate an auxiliary high frequency unit. An electrical filter splits the output of the amplifier, sending low frequencies to the large cone or " woofer ", and high frequencies to

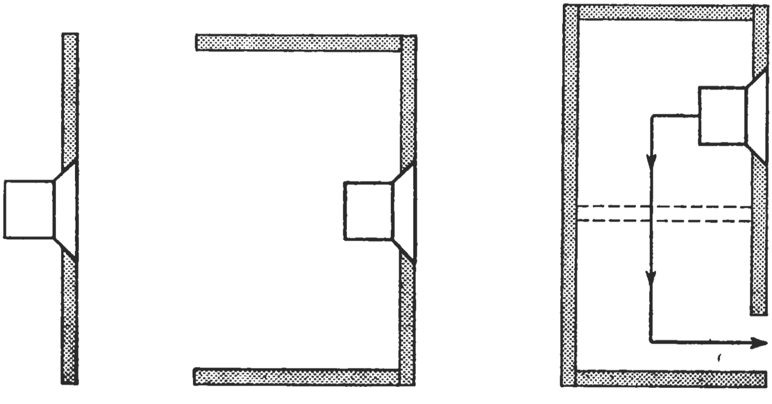


Fig. 6.3. Loudspeaker enclosures: (a) baffle; (b) box cabinet; (c) vented cabinet

the " tweeter ", which may be of the moving-coil type or work on the electrostatic principle described in section 6.2 (Fig. 6.4).

6.2. ELECTROSTATIC LOUDSPEAKERS

Just as there are moving-coil microphones and moving-coil loudspeakers, so the electrostatic microphone has its counterpart in the electrostatic loudspeaker. This principle has been known for many years, and has been used for tweeter units, but until comparatively recently no attempt has been made to produce an electrostatic loudspeaker covering the full audio frequency range. The early electrostatic loudspeakers consisted of a fixed back plate and a conducting diaphragm to which a high polarising voltage was applied, thus creating a charge on the capacitor. The programme voltages were also applied to the capacitor, and the diaphragm would therefore move, reproducing the sound (Fig. 6.5). Unfortunately, as the diaphragm moved in relation to the back plate its distance from it varied, and since the charge on a capacitor supplied at constant voltage depends on the distance between the plates, this varied also. Hence the driving force on the diaphragm varied with amplitude of the signal and so distortion was inevitable. A further difficulty arose in that it was not possible to make the movement of the diaphragm sufficiently free at the same time as preventing it from collapsing on the back plate under the influence of the pull due to the polarising voltage, destroying the capacitor. Such loudspeakers were only possible for use at high frequencies where the amplitude of movement of the diaphragm was sufficiently small to

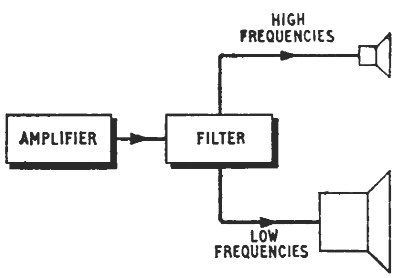


Fig. 6.4. Use of "tweeter"

make the distortion negligible. The small diaphragm movement made it easier to construct a unit that would not collapse.

The recent development of wide range electrostatic loudspeakers has come because of the invention of the " constant charge "

principle (Fig. 6.6). In this case the polarising voltage is applied via a high resistance which tends to prevent fluctuations in charge as the diaphragm moves. The situation can be further eased by mounting the diaphragm between two fixed plates so that the two

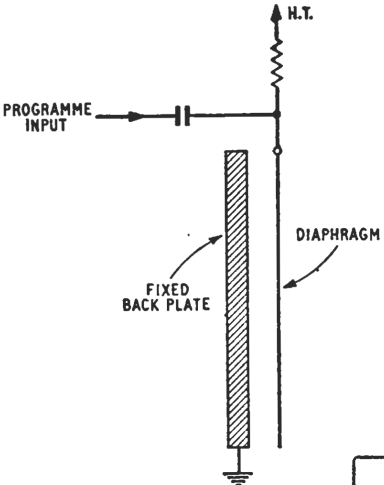


Fig. 6.5. Simple electrostatic loudspeaker

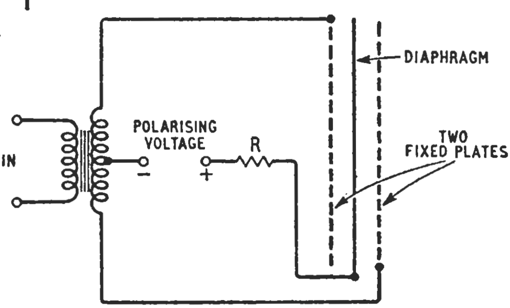


Fig. 6.6. Constant change electrostatic loudspeaker

forces of attraction tend to cancel out. The fixed plates must be perforated in order to allow passage of the sound waves.

As the diaphragm flexes, the gap, and hence the capacitance between it and the plate, will vary, more in the centre than at the outside. It is therefore necessary to make the conducting coating on the diaphragm and fixed plates of a high resistance material to prevent the charge from redistributing itself as the diaphragm moves.

These improvements made the use of bigger diaphragms possible, and loudspeakers of this type are claimed to have wide frequency response, and extremely low distortion. A typical version has a figure-of-eight response at low frequencies and this may be beneficial

in rooms with poor acoustics since the eigentones will not be excited on the dead sides of the loudspeaker. (See Figure-of-eight Microphones, Chapter 5.) There are disadvantages in that a special unit is required to supply the polarising voltage which may be several thousand volts, and a transformer is necessary to match the output of an amplifier, normally of about 15 ohms impedance, to the extremely high impedance of the loudspeaker.

A further disadvantage is that the maximum sound output for a given size of loudspeaker is somewhat less than that obtained from moving-coil loudspeakers of comparable dimensions, and the overall efficiency is lower, due to the difficulty of matching the unit to the amplifier; the electrostatic unit needs more power for the same loudness than the moving-coil one.

6.3. LOUDSPEAKER UNITS IN CURRENT USE

The following loudspeaker units are in current BBC use and since their designs and applications are similar to those used by other organisations they will be described as suitable examples of each type.

6.3.1. LSU/4A

This is a baffle-type loudspeaker of medium quality reproduction used mainly for talk-back. A 10 in. loudspeaker unit is mounted on a 2 ft 6 in.-square baffle and its associated amplifier is mounted on the baffle with its volume control coming through the front. The whole assembly can either be stood on the floor or mounted on a wall bracket. The input impedance of the amplifier is 10,000 ohms and so can bridge a 600 ohm circuit without affecting its level.

Because of the relatively small size of the baffle the bass response is rather poor and so this loudspeaker is normally used for speech.

6.3.2. LSU/10

This is the standard BBC high quality monitoring loudspeaker (see Plate 6.1).

The loudspeaker has two concentric units, and a third unit which radiates extremely high frequencies. The large 15 in. cone is used for frequencies below 1,200 c/s. The second unit has a 1¾ in. diaphragm which radiates via a tapered hole in the magnet pole-piece, through a honeycombed horn. This deals with frequencies up to about 7,000 c/s. The high frequency unit, which is

a separate moving-coil diaphragm unit, with a plastic cone, carries on above this and takes the axial response to well over 15,000 c/s.

The cabinet, developed in BBC Research Department, is of the vented type described earlier. Three layers of carpet-felt stretched horizontally across the centre of the cabinet suppress up-and-down resonances, and together with lagging of the enclosure prevent high frequency radiations from the vent.

The amplifier is a commercial product, and is housed in a shelved compartment at the right-hand side of the cabinet. The input impedance is sufficiently high to permit operation on 600 ohm circuits without loss of level, and the maximum power output, for an input of − 20 dB, is 10 watts.

The height of the open vent above floor-level affects the bass response, and should not be altered—for example, the fitting of castors is not advisable. These were deliberately left out of the original design for reasons of safety, it being felt that such a heavy piece of furniture should be moved only under supervision.

6.3.3. LS3/1 (Plate 6.2)

This loudspeaker, in a totally enclosed cabinet considerably smaller than the LSU/10, was originally designed for high quality monitoring on Outside Broadcasts. It is, however, being increasingly used as a studio monitoring loudspeaker, since its small size makes it easier to accommodate than the LSU/10.

The loudspeaker is used with a separate power amplifier AM8/1, which has a similar input impedance to that of the LSU/10, so that again this can bridge a 600 ohm circuit without affecting level.

The axial frequency response is excellent, extending smoothly to 13,000 c/s, but, due to the relatively small size of the cabinet there is a slight, but not serious, bass loss.

The loudspeaker contains three units—a 15 in. bass unit and two identical high frequency units.

6.3.4. LS5/1 (Plate 6.3)

This loudspeaker is the development from the prototype known as the LSU/12A, described in a paper presented at the Institution of Electrical Engineers in 1958.

It is a studio monitoring loudspeaker of rather larger dimensions than the LS3/1, and has improved frequency characteristics. The power amplifier, AM8/4, is mounted under the loudspeaker cabinet which is of the vented type. The loudspeaker units are the same types as those in the LS3/1 loudspeaker; a 15 in. bass unit and two high frequency units.

6.3.5. Electrostatic Loudspeakers

Electrostatic loudspeakers have been used as monitoring speakers in studio cubicles but are not in general use. It has been found that the maximum sound level produced by existing units has not always been sufficient for studio requirements; further difficulties arise in that the loudspeakers cannot be placed close to a wall, due to their figure-of-eight radiation pattern, and have to be mounted on the floor if loss of bass is not to occur.

6.4. THE IMPORTANCE OF LISTENING LEVELS

The relative sensitivity of the human ear to sounds at different frequencies changes when the level of listening is changed. If, for example, the volume control of a loudspeaker is turned up, the extreme bass and top will be heard to increase more than the middle. Thus, the " balance " of the different frequencies has been upset, in other words we have introduced *attenuation distortion* (see Chapter 12). It is, of course, equally true that listening at lower levels will reduce the high and low frequencies more than the middle and this is often the case under domestic conditions during periods of background listening.

Since the balance of frequencies, as heard by the ear, depends on the listening level, it is important that the studio manager and producer should monitor programmes at the correct volume. How are they to decide what this volume is? Let us look at the different types of programme and see how this decision can be made. In serious music, for example, in the case of a symphony orchestra, the level at the studio manager's ear in the listening cubicle should be as nearly as possible the same sound level as he would hear standing near the main microphone in the studio. This is only strictly true so long as the number of microphones used is few, but nevertheless in multi-microphone balances in this type of music this level will give good results. In the case of chamber music it is seldom that more than one microphone need be employed, so again the sound level at that microphone is the one to choose. It must be borne in mind that many listeners will not be listening at such a high level as this, and checks should be made at a lower level to ensure that the balance remains satisfactory.

Other types of music, however, are not so simple. In the case of dance bands and modern light music, convention and modern musical arrangers decree a balance which may have anything up to 20 microphones, none of which strictly speaking, can be termed the main microphone, and the sound balance in the studio is not

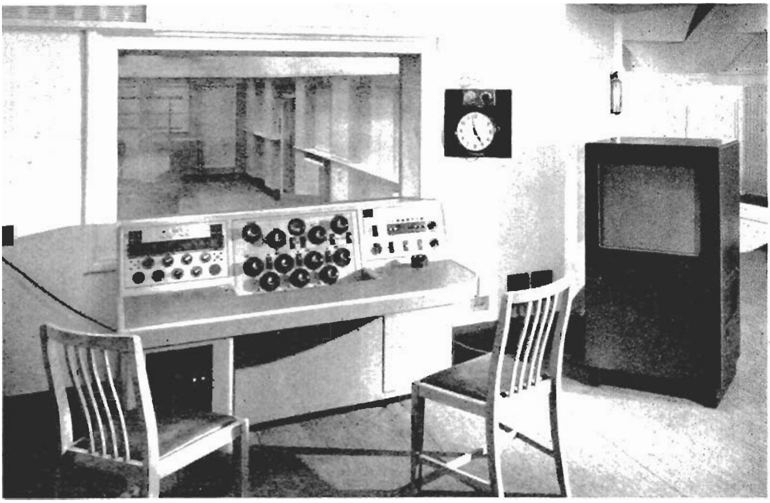


Plate 6.1. Studio control cubicle showing loudspeaker unit LSU/10



Plate 6.2. Corner of studio cubicle showing loudspeaker unit LS3/1

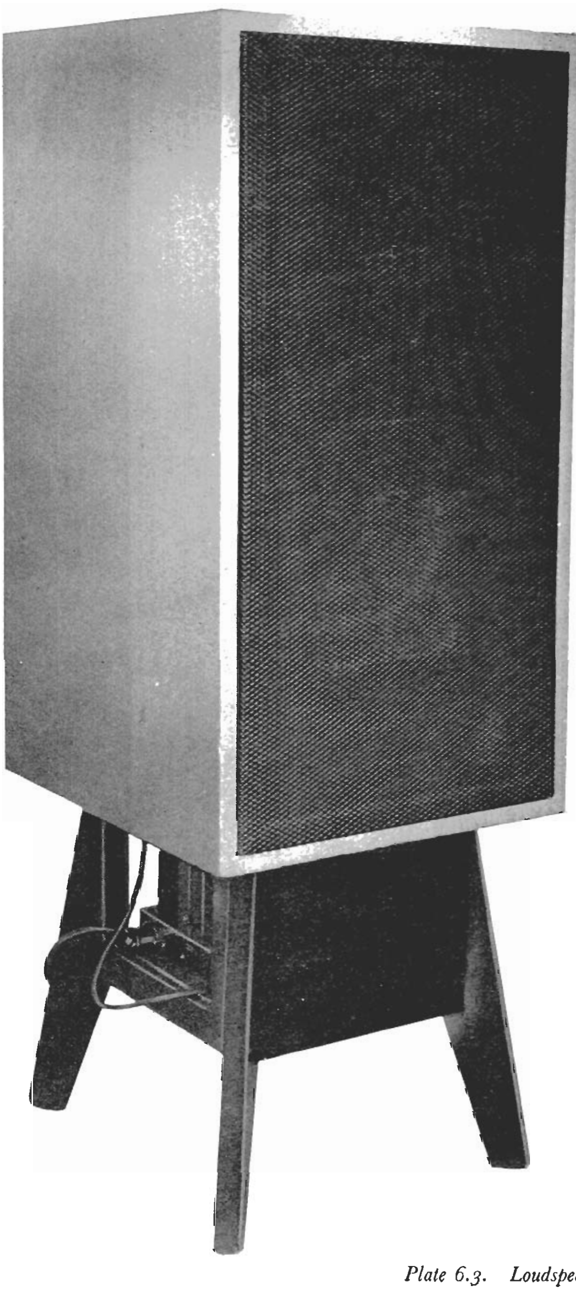


Plate 6.3. Loudspeaker unit LS5/1

necessarily the balance of sound heard in the listening cubicle. In this case the studio manager and producer should set the level of their loudspeaker to give a suitable volume considering the size of the instrumental combination and the type of music being played; again bearing in mind that the average listener may well be listening at a much lower level than this and may therefore lose the effect of some of the high and low frequency components.

A step by step volume control switch can be fitted to the studio desk and recommended settings in decibels are: − 6, −4, −2, normal, +2, +4, +6, +8, +10. The " normal " setting should be adjusted on the loudspeaker control depending on the particular type of programme. The advantage of the switch is the ease with which the volume can be returned to a previous setting when it has been changed for the purpose of checking at a level more nearly that at which an average listener will hear the programme.

6.4.1. Correct Use of the Loudspeaker

The following points should be borne in mind when using a monitoring loudspeaker:

(a) The listening level should be as nearly as possible that which could be heard in the studio as described above.

(b) Frequent checks should be made during rehearsal at a level more nearly that at which the average listener would hear the programme; usually somewhat lower than the level defined in (a).

(c) In the mixing of sound effects and in deciding on the levels for small sounding instruments, such as the clavichord, allowances should be made for the additional background noise on the listener's set.

(d) Monitoring of quality should be carried out near the axis of the loudspeaker, because of the directional effect of high frequencies.

(e) Listening distance should be not less than 3 ft but not greater than 6 ft, since the loudspeaker quality and the apparent reverberation may be seriously modified by the acoustic properties of the listening room. Furthermore, if the loudspeaker has separate high and low frequency units, of necessity some small distance apart in the cabinet, listening too close to

the loudspeaker will not give a correctly integrated sound. Loudspeakers with coaxially mounted units do, of course, not suffer from this difficulty.

(f) There should be no obstruction between the studio manager and the loudspeaker.

7

STUDIO CONTROL DESKS AND ASSOCIATED EQUIPMENT

THE Studio Control Desk provides facilities for mixing a number of sound sources, e.g. microphones, gramophone turn-tables, tape-reproducers and outside sources; controlling their overall volume; monitoring the studio output, and cueing the artists. In this chapter the requirements of such a desk will be described and some examples given from BBC and other equipment.

It is extremely important in this chapter to realize that certain diagrams have been simplified to illustrate separate points. The symbols used to illustrate some components are shown in Fig. 7.1.

7.1. GENERAL REQUIREMENTS

The requirements of any studio controlling equipment can be summarised under seven headings as follows:—

1. Means of mixing various sources:—
   - (a) Microphone sources.
   - (b) Gramophone pick-up sources.
   - (c) Tape reproducers.
   - (d) Outside sources from other studios or organisations.
2. Group and channel switching and control of overall volume.
3. Addition of artificial echo or reverberation.
4. Measurement of programme volume.
5. Means of listening to the studio output.
6. Talk-back system into the studio, and possibly to outside sources.
7. A cue system to the studio, preferably by means of lights.

Each of these requirements will be described in detail.

7.1.1. Means of Mixing Various Sources: Faders

In order to enable the various sources to be mixed in the differing proportions needed to achieve a satisfactory sound balance, each source must be provided with a fader so that its level can be adjusted. Faders are variable attenuators and can take several forms. The simplest form is a variable resistor either in one leg or both legs of the microphone circuit, and until fairly recently many such faders were in use in broadcasting organisations. Fig. 7.2 shows a form

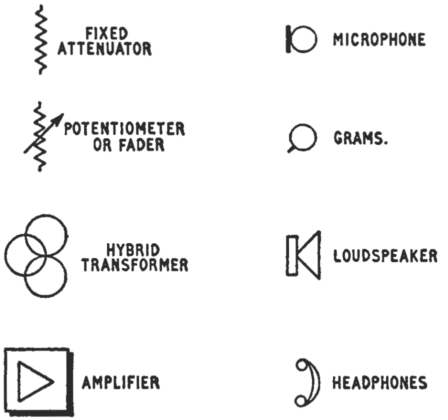
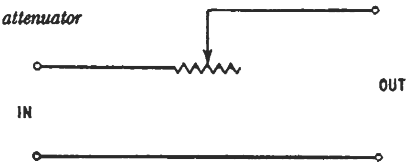
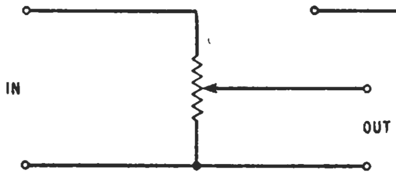


Fig. 7.1. Circuit symbols

of fader often used in domestic and semi-professional equipment—the shunt potentiometer which has many applications. Figs. 7.2 and 7.3 both show unbalanced attenuators, i.e. the resistance is only in one leg of the programme line. A *balanced* circuit is one in which both legs are symmetrically disposed above earth. This form of circuit is often used for carrying programmes—especially at low volume—in order that unwanted currents, such as hum, induced in the two halves by stray electric fields etc. will be identical. Since the currents will be in opposite sense in the closed circuit, they will then cancel. A circuit of this type is shown in Fig. 7.4 with a balanced potentiometer having two sets of sliding contacts ganged together. Both of these simple types of fader, whilst perfectly satisfactory in themselves, can cause difficulties when several are wired together as a mixer (Fig. 7.5). The main difficulty arises because as the fader is moved from minimum to maximum the impedance seen by the following equipment changes considerably, and when the outputs of a number of such faders are combined

Right: Fig. 7.2. Unbalanced series attenuator





Left: Fig. 7.3. Unbalanced shunt attenuator

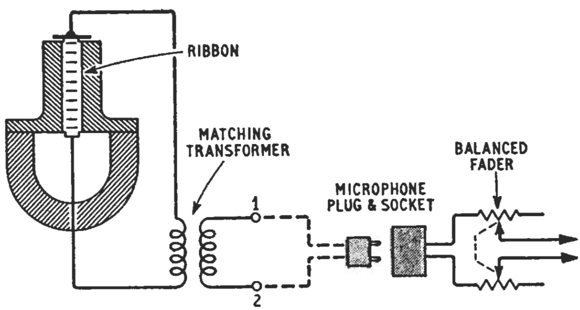


Fig. 7.4. Example of a balanced circuit

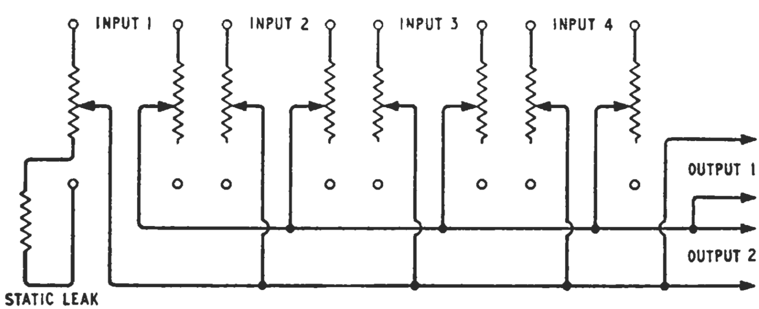
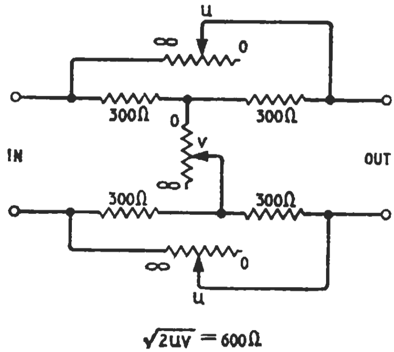


Fig. 7.5. Simple mixer using series faders

Fig. 7.6. Constant impedance fader (600 ohm)



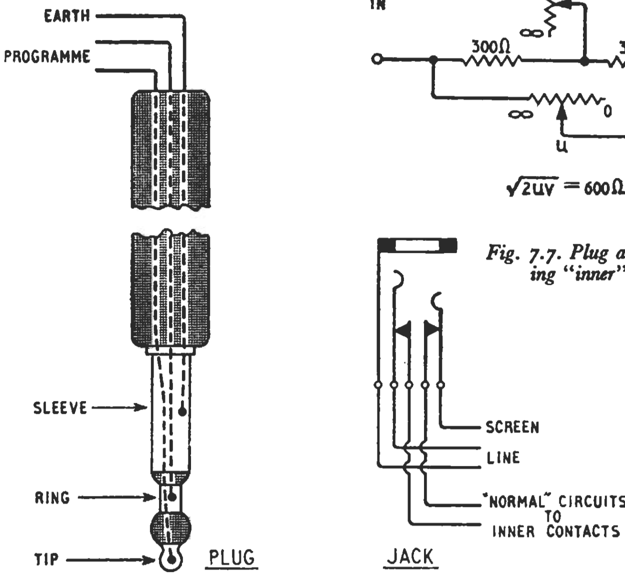


Fig. 7.7. Plug and jack showing "inner" contacts

this impedance change becomes sufficiently magnified to cause a change in overall volume which may make mixing very difficult. The effect, however, can be minimised by never fading the channels up fully, that is keeping some resistance in circuit to act as a buffer between the channels.

A more satisfactory type of fader is the constant-impedance fader which, as its name implies, presents the same impedance to the following equipment no matter in what position it may be set (Fig. 7.6). Such faders are commonly used in professional equipment, but in view of their complex design are normally too costly for domestic use.

7.1.2. Group and Channel Switching

It is obviously desirable, particularly in a permanent studio installation where microphone points are fixed to a wall skirting,

that means should be provided for any given microphone point to be plugged or switched to any channel on the desk. For this purpose plugs and jacks are commonly used and by their means flexible connections in the programme circuits can easily be made. In Fig. 7.7, the plug shown has three connecting points, the tip, ring and sleeve. The programme circuits are connected to the tip and ring and the earthed screening to the sleeve. In the jack inner contacts are sometimes wired so that a complete circuit exists. Inserting a plug breaks this circuit, and the plug circuit then replaces that of the inner contact. This type of " break jack " is used for cross-plugging amplifiers, etc. Some trouble has been experienced at extremely low microphone volumes with circuits made in this way, and it has been found more satisfactory always to plug circuits at this level.

Having arranged the switching of sources to channel faders, it may be desirable to group a number of these channels together to a further fader, so as to be able to adjust them with one control, the

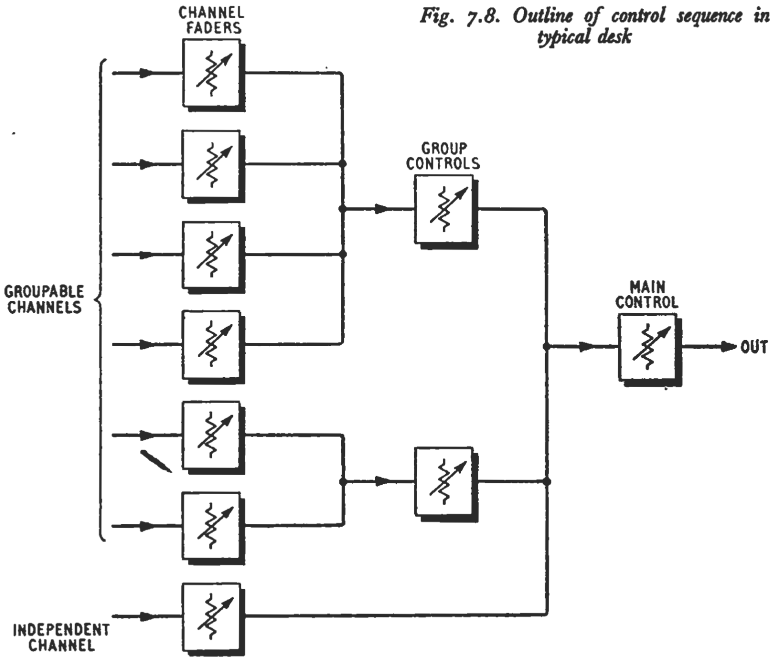


Fig. 7.8. Outline of control sequence in typical desk

remainder of the channels on the desk either being left as individual channels or connected to another group control. One of these individual channels can then be used for a narrator independently of the groups. The outputs of the groups must then be fed to a final main gain control, which is used to set the overall volume of the programme. Switching of channels to groups can again be done by plugs and jacks, or by conventional switches (Fig. 7.8). The main control, since it will be used during transmission or recording to vary the overall volume of the programme, should be graduated in small steps so that no abrupt changes occur as the control is moved. In practice it has been found that with normal programmes, steps of 2 dB are generally considered to be satisfactory. Of course, if the fader is continuously variable as opposed to a stud type, no difficulty arises.

7.1.3. **Artificial Reverberation** (**Echo**)

In studios for music, particularly light music and dance music, and in studios used for dramatic productions, a means of adding artificial reverberation to any given channel is desirable. Control should be provided over the proportion of direct to reverberant

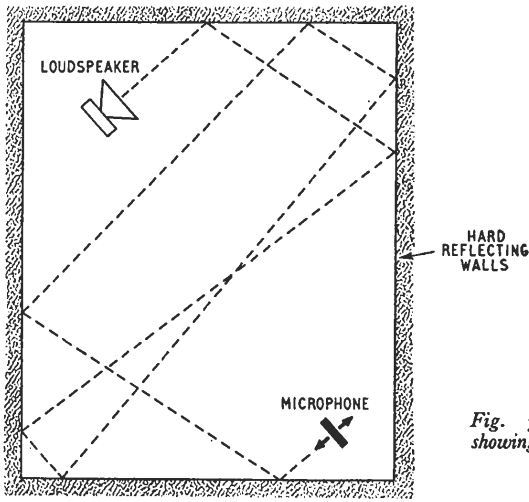


Fig. 7.9. Reverberation room, showing multiple wall reflections

sound on any given channel and an overall control on the output of the reverberation device. The system works as follows:

A hybrid transformer, which is a transformer having one input and two identical and independent outputs, is connected after the

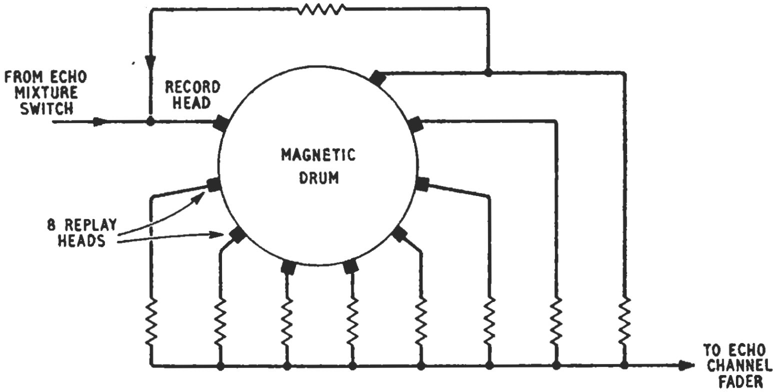


Fig. 7.10. Simplified schematic of magnetic drum reverberation device

channel fader. The two outputs of the hybrid go to the two halves of a ganged rotary " mixture " switch which controls their levels in opposite senses. The hybrid transformer is necessary so that the output of the reverberation device does not feed back via the " direct " side of the mixture switch to the input, thereby causing a howl-round. One output goes to the group fader in the normal way, and the other to the reverberation device. The output of the reverberation device is fed back to the group via a reverberation fader.

Reverberation equipment can be of two types, a room or an electronic device. A reverberation room is simply a bare room having highly reflective walls, in which are placed a loudspeaker and a microphone, the layout being designed so that little direct sound from the loudspeaker can reach the microphone (see Fig. 7.9). The signal from the hybrid transformer and mixture switch is fed to the loudspeaker, and the reverberation produced in the room is fed back via the microphone. Various types of electronic reverberation machines exist. One of these depends upon a magnetic recording of the original sound being played past a number of replay heads, and the output of these heads being fed back into the chain. By this means a number of discrete echoes are produced and by feeding the output of the last of these heads back to the recording head, these echoes can go on, theoretically at least, to infinity (see Fig. 7.10). This device is not too satisfactory since it is often possible to hear individual echoes, particularly when long reverberation times are desired. Another type of electronic reverberation

machine consists of a large rectangular thin steel plate similar in some respects to the traditional theatre thunder sheet. This plate is mounted vertically, tensioned at its four corners. A moving-coil drive unit, similar to that used in a loudspeaker, mounted near one end, causes the plate to vibrate, and a contact microphone, mounted near the other end, picks up these vibrations some time later. The output of this microphone is fed through a suitable equalising

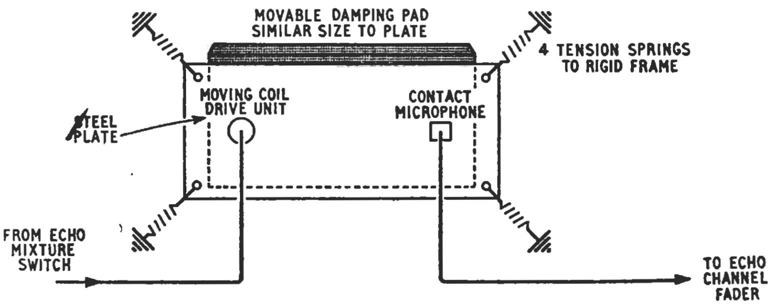


Fig. 7.11. Plate type reverberation device

amplifier back to the programme chain as shown in Fig. 7.11. The equalising amplifier is necessary in order to remove the effects of the various mechanical resonances of the system. Control over reverberation time is possible by moving acoustic damping material nearer to, or farther from, the plate surface. A disadvantage of the plate system is that at the longer reverberation times the bass response tends to rise. This is not, however, serious at the reverberation times most normally used.

7.1.4. Measurement of Programme Volume

In order that the average volume leaving any studio shall be the same as that from another sound source making up a given programme, it is necessary that some form of measurement of the output levels should be made. This measurement will also give an indication when the safety limits of the system are likely to be exceeded either at the high end of the scale, where the onset of distortion must be prevented, or the low end where small signals can cause a poor signal-to-noise ratio. Various devices have been proposed for measuring the programme and two will be described. The first of these, the *volume indicator* or *volume unit meter*, shown

in Plate 7.1(a), is used by many organisations particularly in America. It consists of a rectifier-moving-coil meter having specially designed ballistics. On programme, its readings are arbitrary and the calibration shows " percentage utilisation of the channel ". The scale is calibrated 0–100 with the 100 mark at about two-thirds full scale, the portion of the scale above this being coloured red. The second scale, calibrated in decibels, is for use only on steady tone when the ballistics of the meter can be ignored. This meter is reputed to measure average programme volume, but in fact its readings are purely arbitrary, and it can give misleading results. The reading depends on the type of programme material; a sustained loud programme may well give a lower reading than quieter impulsive sounds. Furthermore, some difficulty is encountered in reading the instrument since the " fall time " of the meter needle is fairly fast, the resulting rapid movement up and down being rather difficult to read. The BBC, therefore, uses rather a different meter, the *peak programme meter* (Plate 7.1(b)). In this instrument, which is a form of valve voltmeter, the circuit is designed so that the meter needle will rise quickly to the peak value of the programme, or very nearly so, and then will fall again slowly. By this means the rapid movement of the needle to and fro, as in the volume unit meter, is avoided and a more even and easily seen indication results. The rise time of the instrument is approximately ·004 seconds and the die-away time approximately 3 seconds. In order to reduce eye strain the scale has been printed white on black and the number of divisions reduced to seven, each division corresponding to a 4 dB change in level.

In BBC practice line-up tone at zero programme level should give a reading of " 4 " on the P.P.M. and this is normally arranged to correspond with a 40 per cent modulation of the transmitter; peaks of " 6 " will then result in full modulation. Peaks above this can cause considerable distortion.

7.1.5. Means of Listening to the Studio Output

It is obviously essential that means should be provided to enable the overall output of the studio to be heard by the person operating the studio desk. This can be by means of headphones, but since the highest possible quality of reproduction is necessary, and since a number of other people may be present, the loudspeaker is essential. Furthermore, since the programme will eventually be heard on a loudspeaker in a listener's home, it is desirable that similar conditions should obtain at the time of balance, since

acoustic conditions may well affect the balance techniques. Whichever form is used it is necessary that it should not cause any loading on the outgoing circuit of the studio, otherwise the P.P.M. levels would be meaningless. Where the P.P.M. is used a convenient output from the P.P.M. amplifier can be taken to the monitoring loudspeaker. The input to this amplifier is arranged to have a high " bridging " impedance.

7.1.6. Talk-back Facilities

A separate system of microphone and loudspeaker is necessary between the studio control desk and the studio itself for communication during rehearsal and transmission. It is convenient for this purpose to use the main amplifiers of the control desk, since in this way talk-back to the studio will also be fed along the output circuit of the studio when this is destined for a recording channel. Safeguards must of course be provided to prevent this happening if the studio is making a live transmission, otherwise the talk-back speech would go over the air. In this condition it should be possible to talk to the studio only, on the headphone circuit, and on the loudspeaker when the microphones are faded out. Talk-back to outside sources, where these are used, may be desirable and this would normally be carried along the control line associated with this circuit.

7.1.7. Cue System

A number of cue lights in the studio, switchable from the control desk, should be provided, preferably one for each microphone point. It is convenient if these can be operated by a key adjacent to the fader of the microphone concerned. This will necessitate switching for cue lights in order to select key positions, since the microphone point in the studio will already have been plugged to a particular channel.

7.2. EXTRA FACILITIES ON BBC CONTROL DESKS

7.2.1. Programme Ring-main Switch (HV/7)

When a studio is about to join or leave a particular programme service, it is desirable for it to be able to hear the preceding and following contribution. For this purpose a twelve-way ring-main is provided in all BBC studios.

A key, normally coloured white, is provided to enable a quick

switch of the monitoring loudspeaker to be made from the ring-main point selected to the studio output (Fig. 7.12).

7.2.2. Clean Feed

It may be necessary, when working with outside sources, to provide a *clean feed* of the programme leaving the studio, to be fed back to the outside source. By " clean feed " is meant everything passing through the studio desk, except the contribution from

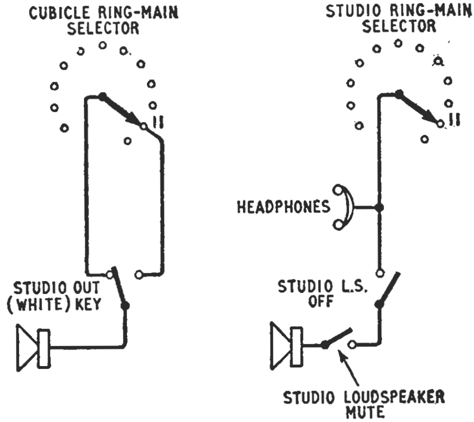


Fig. 7.12. Programme ring-main switches (HV/7). LS mute switch operates in studio when microphone is faded up

the source to which the clean feed is sent. Two reasons make this necessary. If the contributing source is a great distance away, say the other side of the Atlantic, it will need to hear the cue programme from the BBC studio, say on a pair of headphones or even a loudspeaker. If, when the BBC switches over to his contribution, the speaker hears his own voice back, there can be a long distance howl-round if a loudspeaker is in use, and if he is listening on headphones his own voice will appear with an appreciable delay, due to the length of the line. This delay makes normal speech delivery rather difficult, so under these conditions clean feed is desirable. The second use for clean feed is when the remote contributor is broadcasting simultaneously in his own country both his speech and the output of the BBC studio, a genuine two-way programme; in this case the need for prevention of howl-round is obvious.

7.2.3. Public Address

In studios where audiences are present, it is often necessary to provide feeds for public address equipment from one or more of the microphones being used for transmission. This is done by means of hybrid transformers in the microphone circuits and a

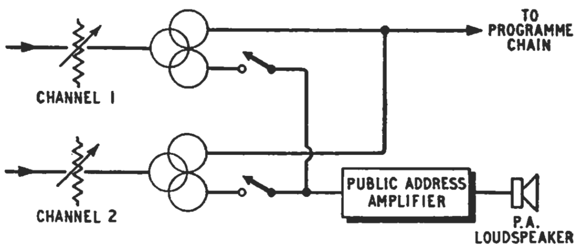


Fig. 7.13. Switching of individual channels to Public Address

series of switches, one per channel, to enable the microphones required to be switched to the public address (P.A.) amplifier. Two positions of each switch give normal level and + 10 dB feeds, and an overall P.A. level control is also provided (Fig. 7.13).

7.3. TYPE A STUDIO EQUIPMENT

7.3.1. Type A Studio Desk—General

The complete Type A console comprises a control desk, a studio control cabinet, and a studio supply cabinet.

(a) *The Supply Cabinet* houses the mains isolators and contact-breakers.

The equipment isolator has three positions:—

I. Remote Control—when mains supply is fed via a " Mains On ", relay operated by the red, green and orange push-buttons on the desk.

II. Off.

III. Direct Control—when mains supply is fed direct to the equipment; this position is to be used in the event of failure of the 50-volt relay supply.

(b) *The Control Cabinet* (see Plate 7.2) houses all amplifiers and relays, and a pre-mixer jackfield for cross-plugging purposes. Access to the jackfield is possible by a small door at the bottom right-hand corner of the cabinet.

(c) *The Control Desk* has the various controls mounted on three upright panels as follows (see Plate 7.3):—

Left-hand panel:

- Amplifier change-over keys
- Local red light key
- Push-buttons, for ON/OFF, etc.
- Standby loudspeaker key
- Monitor to line key
- Echo mixture switches (if required)
- Echo selector key, when alternative or shared echo facilities are available
- Clean-feed selector key

Centre panel:

- Mixing and control potentiometers
- Programme meter
- Cue light keys (and indicator lamps)
- Echo-cut key (if required)

Right-hand panel:

Top row:

- Control-room buzzer

Middle row:

- Loudspeaker dim key
- Telephone I ring/recall key with white indicator light
- Remote record key with red indicator light
- Two outside source groups, each containing:
  - Call button
  - Pre-fade listening button
  - Recall/answer key
  - Cue to control line or cue line key
- Programme selector key giving pre-fade or studio output on loudspeaker

Bottom row:

Three local tape record/replay keys
Clean-feed talk-back key
Studio talk-back key
Two outside source groups if fitted
Studio loudspeaker ON/OFF switch

In addition:

Acoustic effects volume control
Cubicle loudspeaker volume control
Studio programme selector switch
Cubicle programme selector switch
Two pre-fade headphone jacks

The left-hand telephone compartment also houses the standby loudspeaker, and the headphone jacks, which are connected to the " studio " programme selector switch.

Two basic forms of Type A Console are in use, with the following facilities, which can be built upon as required:—

Mark II—5 channels.

Mark V—7 channels or sometimes 9, echo facilities, group and independent faders.

There is a single Mark VII console in Piccadilly Studio I, providing 12 channels, and special arrangements for grouping of sub-channels. The notes which follow refer to the Standard Mark II and V equipments.

7.3.2. Circuit Details (Fig. 7.14)

The Type A equipment was introduced shortly after the war, and incorporated a number of novel features:—

(a) Constant-impedance faders.
(b) Individual microphone amplifiers.
(c) Cross-plugging of sources and channels.
(d) Push-button selection of rehearsal, line-up and transmission conditions.
(e) Comprehensive echo facilities (if required).
(f) Standby amplifiers and change-over keys.
(g) Arrangements for clean-feed working.

These will be considered in order.

(*a*) *Constant-impedance Faders*

All the potentiometers on the Type A desk are designed to present the same impedance (600Ω) in and out. The circuit is as shown in Fig. 7.6.

The *microphone* and *group faders* have twenty steps of 2 dB—increasing to 8 dB at the lower end. Fading out also feeds programme to the studio loudspeaker, by means of an extra contact

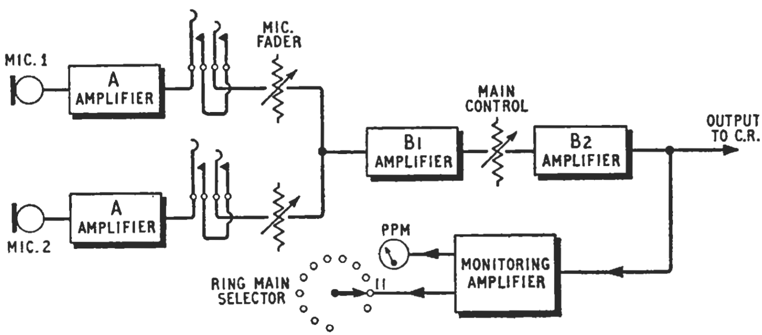


Fig. 7.14. Simplified diagram of Type A Mark II

on the last stud. A faulty fader may be removed and another substituted in a few seconds. But note that in certain studios, where extra mixers have been installed, the group faders are of a special type and cannot be interchanged with standard faders.

The *Main Control* fader has 30 steps, from 2 dB to 5 dB at the lower end.

(*b*) *Individual Microphone Amplifiers*

The introduction of constant-impedance faders, as just described, plus a special circuit of fixed resistors, ensures that accurate matching of a given number of channels is maintained, however many of the channels are faded up. This eliminates the drop in programme volume which accompanies fading-in additional channels in older equipment, but involves an inherent fixed loss of about 30 dB.

Such a loss would be very serious on programme at microphone volume, because of noise, and pre-amplification is therefore necessary. For this reason, an amplifier is included in each microphone circuit and the gramophone circuit, before the fader. The gain of

this " A " amplifier is 50 dB, so that mixing is carried out at about — 30 dB instead of — 80 dB.

Following the mixing circuits are two control amplifiers, B1 and B2, between which comes the main control potentiometer. The output of the B2 amplifier is at zero volume, and feeds the programme line to the control room. The B2 also connects to the programme selector switch point 11 and the programme meter, via a monitoring amplifier.

(c) *Cross-plugging of Sources and Channels*

The outputs of the A amplifiers are taken to the channel faders via a jackfield in the control cabinet. This permits cross-plugging

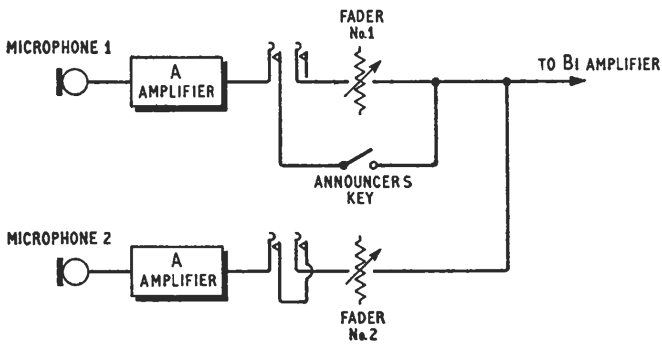


Fig. 7.15. Action of announcer's key in news studio

of any source—mic., gram., or outside source—to any fader, to simplify operation.

An exception to this exists in the News studios. Mic. 1 in this case is normally connected to the control amplifiers through an announcer's microphone key, and not via a channel. If mixing is necessary, Mic. 1 is plugged to a channel, the " censor " key then being inoperative (Fig. 7.15).

(d) *Push-button Selection of Conditions*

Rehearsal Conditions: Pressing the green button operates the " Mains On " relay, and provides full talk-back facilities. Operating the talk-back key then connects the talk-back microphone to the control B1 amplifier, short-circuits the main control potentiometer, cuts the cubicle loudspeaker, and feeds the headphones and studio

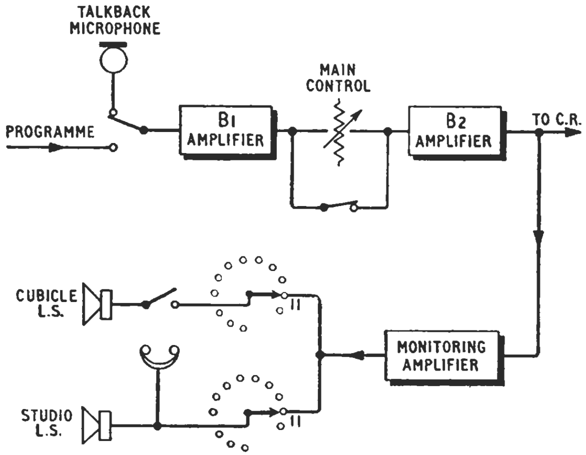


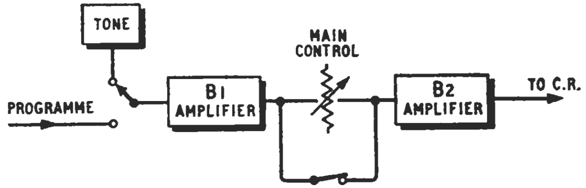
Fig. 7.16. Action of talk-back key in rehearsal condition

loudspeaker, whether microphones are faded up or not. Note that talk-back volume does not depend on the main control setting, and that the studio loudspeaker programme selector must be at point 11 (Fig. 7.16). (If the selector is switched to a cue programme, this will be heard in the studio when the talk-back key is pressed.)

Line-up Condition: Pressing the orange button operates the "Mains On" relay, applies 1,000 c/s tone to the input of the B1 amplifier and short-circuits the main control potentiometer. The tone has been suitably attenuated to give zero level into the programme line. This level does not depend on the setting of the main control potentiometer, and provides a check on the gain of the control amplifiers (Fig. 7.17).

Transmission Condition: Pressing the red button operates the "Mains On" relay, and provides restricted talk-back facilities

Fig. 7.17. Line-up condition



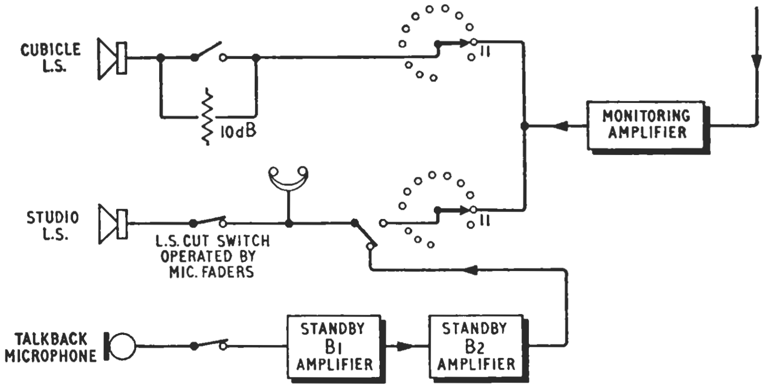


Fig. 7.18. Action of talk-back key in transmission condition

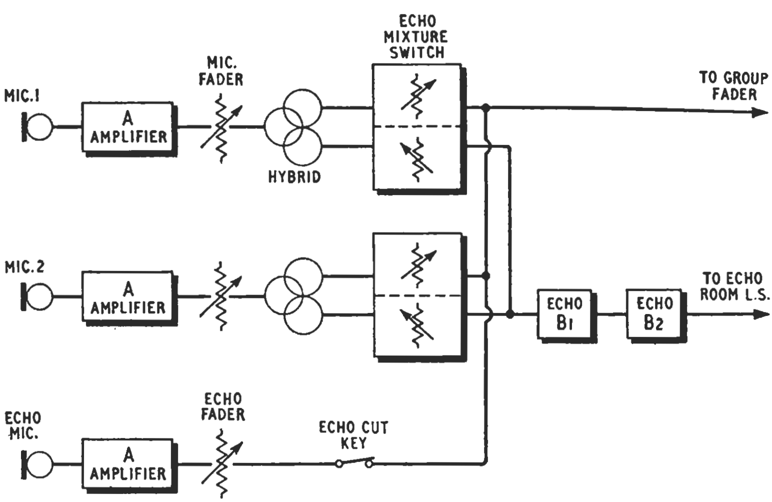


Fig. 7.19. Derivation of artificial reverberation feed in Type A Mark V

(Fig. 7.18). Operating the talk-back key then connects the talk-back microphone to the *standby* B1 and B2 amplifiers, dims the cubicle loudspeaker, and feeds the headphones and studio loudspeaker—via the loudspeaker cut relay operated by the microphone faders. Note that talk-back is at fixed volume, does not enter the programme line and appears on the studio loudspeaker only if all microphones are faded out. In this case, the studio loudspeaker programme selector need not be at point 11. Talk-back is automatically fed to headphones, as in rehearsal conditions. A toggle switch is available to turn off the studio loudspeaker, if required, and when this is done talk-back is only available on headphones.

Transmission condition is also set up and held by the signalling red light voltage from the control room. In order to provide full

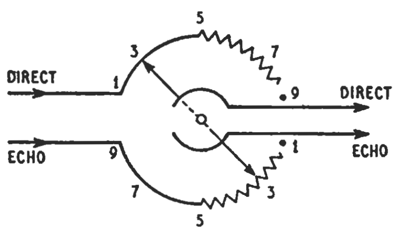


Fig. 7.20. Reverberation mixture switch

talk-back facilities it is usual to carry out recordings in the rehearsal condition, using the local red light key to operate the warning lights outside the studio and cubicle doors.

(*e*) *Comprehensive Echo* (*Reverberation*) *Facilities*

On Mark V and larger consoles, it is possible to pre-select echo on all channels, except the Independent channel. Coarse selection of the direct/echo ratio is made by the echo mixture switch for the channel(s) in question. Fine control of the amount of echo is possible on the echo microphone fader. An echo cut key permits instantaneous " make " or " break" of echo.

The chain of events may be traced by reference to Fig. 7.19 and the following description:—

The Echo Chain. The output of each channel fader is split into two separate feeds by the *hybrid transformer*, which has two identical and independent secondary windings. These are connected to the direct and echo halves of the echo mixture switch (Fig. 7.20). This

is really a two-channel fader, introducing loss in the direct and echo chains on clockwise and anti-clockwise movement respectively.

In position 5 of the switch, no attenuation appears in either chain, and the amounts of echo and direct are equal. Switching from 5 to 1 reduces echo while leaving direct at full volume. Conversely, advancing from 5 through to 9 leaves the echo full on and progressively attenuates the direct until at position 9 the direct circuit

Table 7.1

DIRECT AND ECHO LOSS IN EACH CHANNEL

Position	*Direct*	*Echo*
1	0	Infinite
2	0	12 dB
3	0	7½ dB
4	0	3½ dB
5	0	0
6	3½ dB	0
7	7½ dB	0
8	12 dB	0
9	Infinite	0

is broken altogether. The loss in each channel for all switch positions is given in Table 7.1.

The *direct* outputs of all switches are connected together, and taken to the group fader. The *echo* outputs are taken to the *echo room loudspeaker*, via echo B1 and B2 amplifiers.

The *echo microphone* picks up this output, plus reverberation, and feeds it back to the group fader, via A amplifier, echo microphone fader, and the echo cut key. Howl-round of the echo chain is prevented by the hybrid transformer.

The cross-plugging jackfield comes between the A amplifier and the fader, as usual, so that equipment for modifying the frequency response may be inserted. The output of the echo microphone cannot be cross-plugged to any other channel. The *independent*, or *narrator's* channel cannot have echo, and bypasses the group fader into the B1 amplifier.

(*f*) *Standby Amplifiers*

As indicated in the simplified diagram (Fig. 7.21), standby amplifiers are provided. There is an amplifier change-over key

for each A amplifier. Throwing one of these keys down replaces the appropriate A amplifier by the X standby amplifier. Throwing a key up substitutes the Y standby.

If channels are cross-plugged, the change-over key associated with the *source* not the channel, must be selected, engraved studs being provided for insertion alongside appropriate faders.

A further change-over key substitutes standby B1 and B2 amplifiers for either the control (key down) or echo (key up) B amplifiers. Since transmission talk-back (see section (d)) uses the spare B amplifiers, this facility is carried by faulty amplifiers if the spares have been selected to replace faulty control or echo amplifiers.

If the *cubicle loudspeaker* fails, and no headphones are available, the key labelled " *standby loudspeaker* " may be thrown, and connects the L/S in the left-hand telephone recess direct to the HV/7.

If the *monitoring amplifier* fails, the key labelled " *monitor to line* " may be thrown, and connects ring-main point 11 direct to the

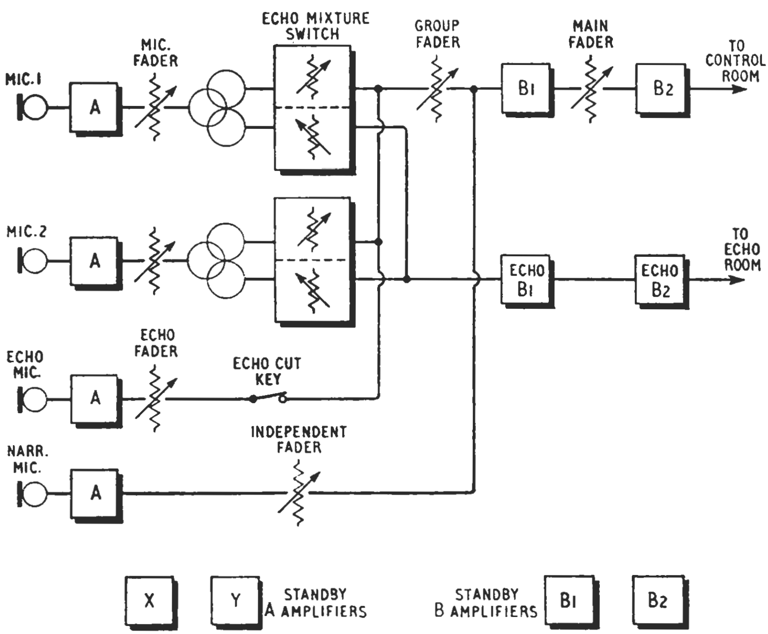


Fig. 7.21. Simplified diagram of Type A Mark V

output line (Fig. 7.22). Complete failure of the monitoring amplifier would, of course, render the programme meter inoperative.

(g) Clean-feed Working

When a programme from a given studio includes contributions from remote studios or O.B. points, it is usual to feed the output of the main studio desk to the outside sources for cueing purposes

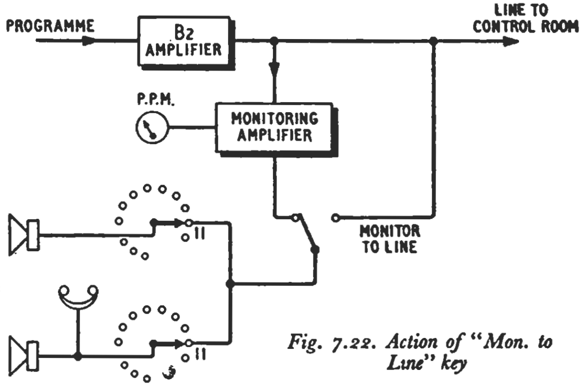


Fig. 7.22. Action of "Mon. to Line" key

and talk-back. In the ordinary way this cue feed will consist of the complete programme as mixed at the main studio.

In clean-feed working this may be used by the remote studio as a source of programme, mixing taking place at both ends. It is therefore essential that the remote contribution is not included, since howl-round might result. This type of feed has also been found useful for public address and cueing over long circuits when the remote contributor has not wished to hear his own words on headphones.

Type A desks can have up to four outside sources, one of which is equipped for clean-feed working. (In some exceptional studios, all Outside sources are equipped for clean-feed working if required.) The locking key on the left-hand panel labelled " clean feed " and " clean-feed and TTB " gives this facility as required (Fig. 7.23).

In the centre position of the key (normal), no clean feed is provided, communication and cueing of outside sources being effected by means of the control line, and the remote contributor will hear his voice coming back.

In the clean feed position of the key, used for genuine two-way working, the output of the studio, except for the outside source

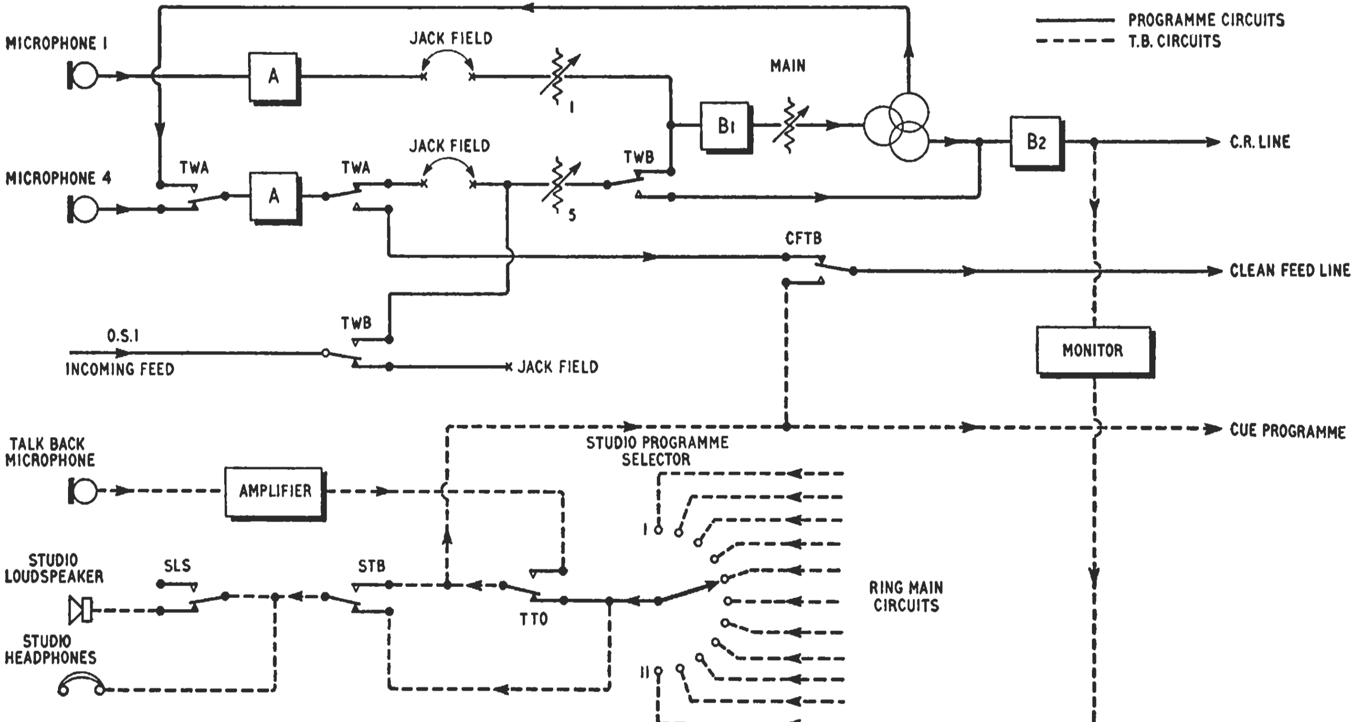


Fig. 7.23. Clean-feed working. All relays shown in "Normal" position. When clean-feed key is made *relays TWA, TWB operate transmission talk-back available to studio and cue circuit only via TTO, STB.* When clean-feed and TTB is made, *TWA, TWB operate as before transmission talk-back is available to studio and cue circuit via TTO, STB as before, and also, on separate key, via CFTB, TTO to clean feed*

contribution, is fed back along the clean feed line to the outside source. In the rehearsal condition studio talk-back will also go to the outside source, but in the transmission condition this is undesirable since it would be radiated by the distant station.

In the "clean-feed with Transmission Talk-back" position of the key the conditions are as above with the addition that it is possible to talk to the remote contributor under transmission conditions by using the "clean-feed and talk-back" key. This facility is useful when the remote contributor is at a great distance away, say in another country, when it would be undesirable for him to hear his contribution " coming back " as in " Normal " working. The time delay inherent in long land lines would be very confusing for him since he would hear each syllable a fraction of time after he had uttered it.

Points to remember in clean-feed working are:—

(1) The incoming feed is tied to a particular channel fader and must not be cross-plugged.

(2) The incoming feed does not pass through the main fader, and care must be taken to fade out the channel as well as the main fader at the end of a recording or transmission. Also, since the level of the incoming contribution cannot be boosted by means of the main fader, it should be checked at rehearsal, and cases of low level referred to the control room.

(3) The facility should be used only in cases where a clean feed is required—e.g. in exchange programmes with the Continent, etc.—and the key should remain in its Normal position for ordinary programmes involving outside sources—O.S.1, O.S.2, etc.—which can be cross-plugged in the usual way to any channel.

7.4. MARCONI STUDIO CONSOLE (Plate 7.4)

A number of Talks studios in London and the Regions have been equipped with commercial desks of this type. They provide high level mixing of four channels. The normal arrangement is shown in Fig. 7.24—three low-level studio sources, including grams, and an outside source channel to which one of a number of outside sources may be switched. The faders are numbered in the diagram to show their position on the desk, reading from left to right.

A headphone jack and " monitor " switch on the desk permits pre-fade listening on all channels and outside source lines—points marked " X " in the diagram.

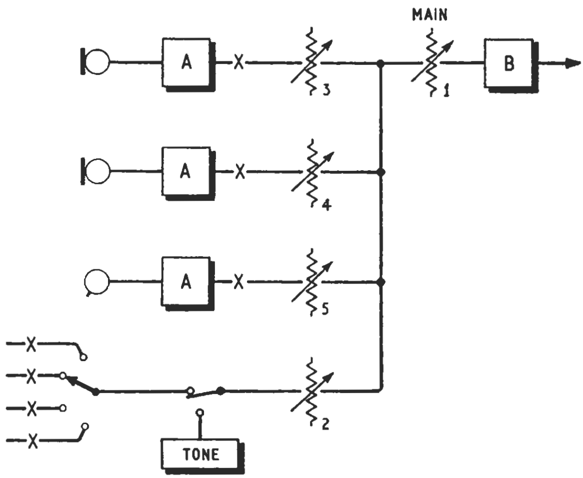


Fig. 7.24. Circuit of Standard Marconi Desk

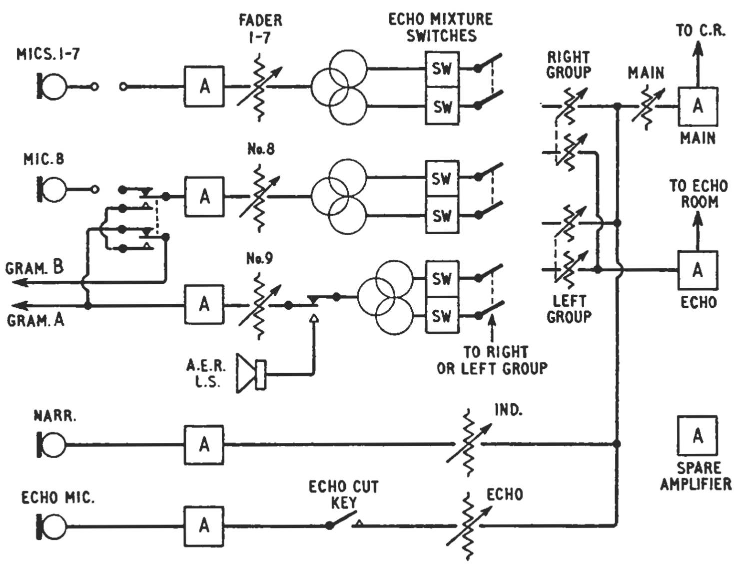


Fig. 7.25. Type B Studio Equipment (Mark II)—simplified diagram

Other points of difference from the Type A Equipment may be summarised as follows:—

1. No standby amplifiers.
2. Rehearsal talk-back bypasses the programme selector switch.
3. No transmission talk-back.
4. Line-up tone is usually fed through the O.S. fader (No. 2) as well as the main fader (No. 1).
5. The signalling red light from the control room holds the equipment in transmission condition (to prevent accidental use of talk-back) as long as it is applied, and the equipment reverts to rehearsal condition, when the light goes out.
6. The faders cannot be removed from the panel.

When a third microphone is provided in the studio, it is wired as one of the O.S. inputs to fader No. 2.

7.5. TYPE B STUDIO EQUIPMENT

7.5.1. Type B Studio Equipment—General (Plates 7.5, 7.6 and 7.7)

A feature of the control desk is the use of standard $5\frac{3}{16}$ in. long by $4\frac{1}{4}$ in. wide panels. It was planned, on the " building bricks " principle, to treat each installation as an individual for the detailed grouping of these panels. A principal feature of the equipment is its flexibility, enabling a great variety of circuit connections to be made. For example, provision is made for clean-feed working.

7.5.2. Circuit Details—General Purpose Desk (Mark II) (Fig. 7.25)

The Standard Type B desk will carry up to 10 channels with a possible extension to a maximum of 12. The facilities provided will be taken in order.,

(*a*) *Source Selection* (Plate 7.6)

Source to channel selection is possible, but no automatic connections exist. It is necessary to use a double-ended cord for *every* connection. Notice also that cross-plugging takes place *before* the pre-amplifier (unlike Type A), so that exchanging channels is an effective cure for a faulty amplifier.

Channel 9 is wired permanently to grams, and it is possible to switch its output to an *acoustic effects* loudspeaker in the studio. A toggle switch alongside channel 8 can be thrown to split grams

between channel 8 and 9. In this case the channel 8 cannot be used for a microphone. The switch alongside channel 9 will then connect the turntables left to it either to Direct or Acoustic. The actual division of turntables between channels 9 and 8B is fixed at the time of installation, but in a 6-turntable studio it would usually be 4 turntables on channel 8B and 2 on channel 9.

(*b*) *Group Selection*

There are two group faders, and all sources except the independent channel 10 may be connected to either group by means of a *group switch* on the channel panel. Blue and red indicator lamps show whether a channel has been selected to the right or left group respectively.

(*c*) *Cue Light Selection* (Plate 7.6)

Each of the green cue lights in the studio may be switched on a cue selection panel to any or all of the cue keys which are fitted beside every microphone fader. The operations of fading up and cueing can thus be performed with one hand.

(*d*) *Echo Selection*

Echo is available on all channels except the Independent, by means of the type A arrangement of individual channel selection, the echo mixture switches being on the left of the main panel.

(*e*) *Standby Amplifiers*

A spare amplifier, which carries transmission talk-back, may be interchanged with either the control or echo B amplifier. Inserting a spare A amplifier involves setting a rotary switch on the amplifier bay to the appropriate position (when the faulty and spare amplifiers are operating in parallel), and then physically removing the faulty amplifier.

(*f*) *Distortion Units*

Two effects units are wired to preferred channels, one of which will always be grams, so that bass cut and top cut may be inserted by switches on the panel.

(*g*) *Public Address Selection*

Where public address loudspeakers are required, a special panel of switches selects channels 1 to 9 to be fed to the P.A. amplifier.

(*h*) *Pre-fade Facilities*

Outside sources are provided and pre-fade listening to these sources is selected by push-buttons. The panel which carries the

pre-fade buttons also gives flexible cue speak facilities. The control line normally carries cue programme and talk-back to the outside source. This is replaced by the *control telephone* when the *speak key* is depressed.

(*i*) *Push-button Selection*

Buttons are provided, as with Type A, for rehearsal, line-up, and transmission. No locking into transmission condition by the red light voltage occurs.

(*j*) *Clean-feed Facilities*

The incoming contribution is fed to the *blue* group and all studio sources must be fed to the *red* group. The *blue* group control then acts as the fader for the source to which clean feed is sent. Clean feed, and clean feed with TTB, are provided as with Type A.

7.5.3. Talks and Special Desks

(a) The *Talks and Discussions* version of the Type B desk (Mark I) is accommodated on a panel of the same dimensions as the centre panel of the General Purpose desk. The bottom row of five channels are, from left to right, microphone 1, microphone 2, incoming contribution, tape and gramophone. The incoming contribution channel is for use in clean-feed working only. The middle row of panels carries the main control, and two outside source faders, extending on certain installations to a maximum of four.

In general, the facilities are the same as with the General Purpose desk, except that no echo, group, or independent channels are provided.

(b) The *Special* versions of the desk (Mark III) have been designed to suit individual large studios. A typical arrangement which is regarded as a working maximum consists of 11 main channels on the centre panel, together with the main control, group, independent and echo faders, plus a further pair of independent channels, and 16 sub-channels mounted as two 4-channel mixers on either side of the centre panel. One or more of these mixers may be plugged to any of the 11 main channels, or 3 Independent channels, for grouping purposes. The maximum number of sources available simultaneously is 28. Amongst other facilities found in the Mark III equipment are the following:—

1. Switchable cue lights up to a maximum of 11
2. An alternative echo source, when available, necessitating altering the Independent channel on the centre panel to " Echo 2 ",

and provision of a switch alongside each Echo Mixture Switch to select the two echo rooms or machines

3. Cubicle L/S may be switched to monitor the public address—e.g. during the sending of " line-up " tone when an audience " warm up " is in progress
4. Public address switches for all eleven main channels, and 1 Independent channel—having an up position giving 10 dB boost. It is also possible to feed echo 1 to P.A.
5. Distort units pluggable to any channel
6. Split gram outputs pluggable to any channel

7.6. MIXER SUITE B.1, BROADCASTING HOUSE

Detailed descriptions of particular installations are outside the scope of this handbook, but the B.1 mixer suite in Broadcasting House incorporates a number of interesting features which may indicate future trends in design, and therefore a short description may be useful.

As the photograph, Plate 7.8 shows, the mixer desk is based on the Type B studio equipment, with special arrangements to suit the primary function for which the suite was designed—namely, the mixing and direction of large-scale broadcasts involving a number of outside sources or studios.

7.6.1. Mixing Facilities

The principal mixing controls are situated on the centre panel of the mixer desk, with the main control fader on the left-hand panel. A programme meter is provided on each of these panels, in case a second studio manager becomes necessary to control the overall volume. There are two *group faders* at the top of the centre panel. Each controls the row of three channel faders immediately below it, coloured red and blue respectively. Each of these channel faders may be used to control any one of four different sources by push-button selection. All sources are plugged to the channel inputs on the control position in the engineer's cubicle, in accordance with the layout requested by the studio manager, and selection of the required source on each four-source fader is achieved by pressing the appropriate push-button on the left-hand side of the fader. When connection is made of the source to the input of the fader, the corresponding lamp above the fader will light.

This selection of a source is possible only when the channel is faded out. Thus, if source L.G.1A is faded up (the first source on

the first fader on the left group), the button associated with L.G. B, C, or D may be depressed, but the connection to the input of the fader will not change over until the fader is faded out. This system of pre-selection is designed to prevent accidental change-overs, and to facilitate the change-over when consecutive programme items come up on the same fader. When the running order is known in advance, of course, sources which follow consecutively should be plugged to different faders, and when the total number of sources is 10 or less, each should be allotted a separate fader.

Four *independent channel faders*, designated W, X, Y, and Z, occupy the bottom row on the centre panel. The desk as a whole can therefore be used with up to 28 sources, although only 10 channel faders are provided.

7.6.2. B.1 Mixer Local Output

The B.1 mixer room includes an announcer's desk of the type employed in B.H. Continuity studios. The output of this desk can be plugged as a source to the mixer desk, and includes turntables for 78 r.p.m. and microgroove discs, and a microphone suspended over the announcer's position. This is intended to link composite programmes through the mixer, or can be used as a standby continuity suite. The outputs of the three 78 r.p.m. desks and the L.P. desk also appear as part of the mixer output.

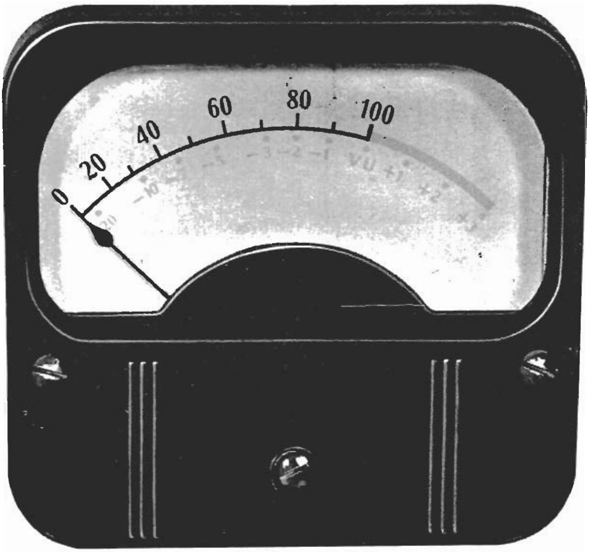
An auxiliary microphone socket and fader on the right-hand side of the mixer control desk are intended for use with a lip microphone. It is important not to fade up the announcer's desk microphone simultaneously with the lip microphone, since the bass filter circuit of the lip microphone will deteriorate the quality from the other. When either microphone is faded up, the mixer room loudspeaker is automatically cut or dimmed in the ordinary way.

7.6.3. Local Tape Facilities

A special jackfield is provided on the left-hand wall, whose 28 jacks, suitably coloured and labelled, are connected in parallel with the 28 channels on the desk. By connecting the appropriate jack to a recording machine, it is thus possible to continue radiating from any channel, while recording from another for future reproduction.

The output of one or more tape machines set up in the mixer may be plugged as a source to any channel of the desk. The correct procedure for this is to plug the output of the tape machine to the *repro.* jack, where it can be picked up on the Control Position, and

(a)



(b)

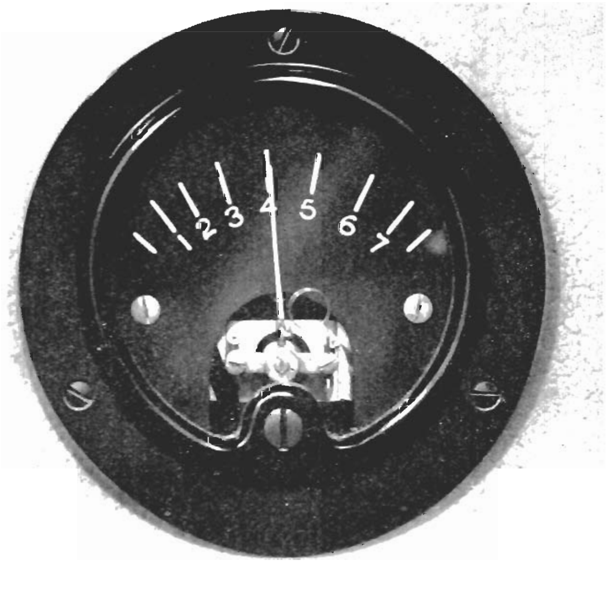


Plate 7.1 a . VU meter. b Peak programme meter

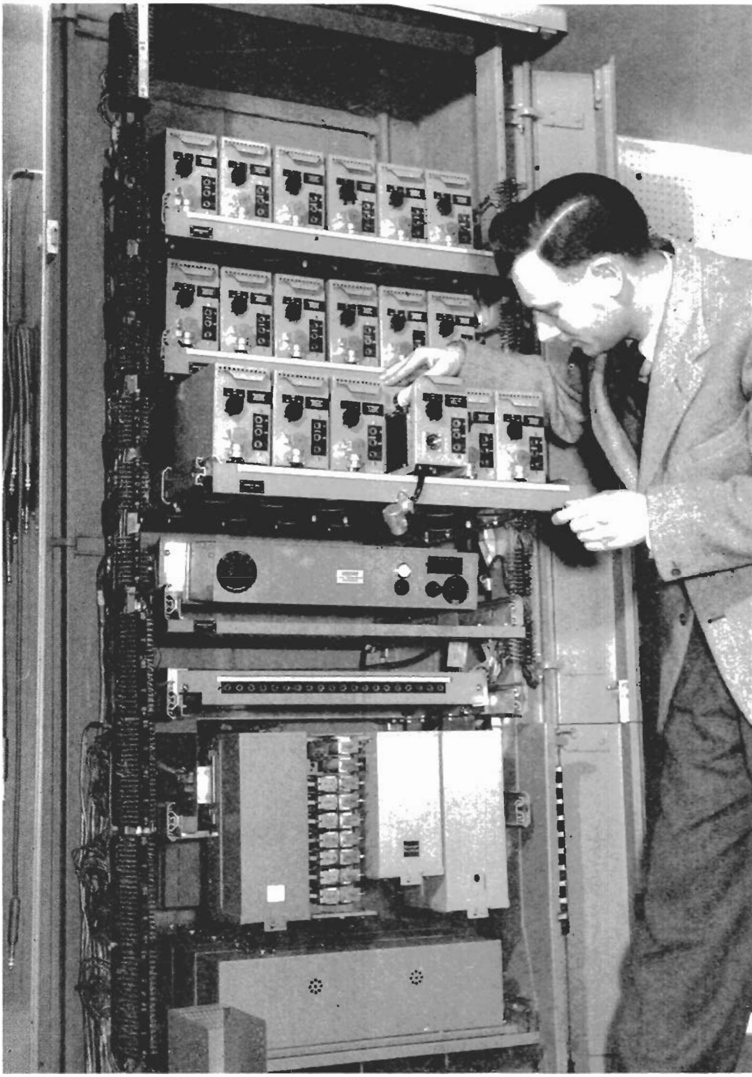


Plate 7.2. Type A control cabinet



Plate 7.3. Type A console



Plate 7.4. Studio control console Marconi Type BD.545

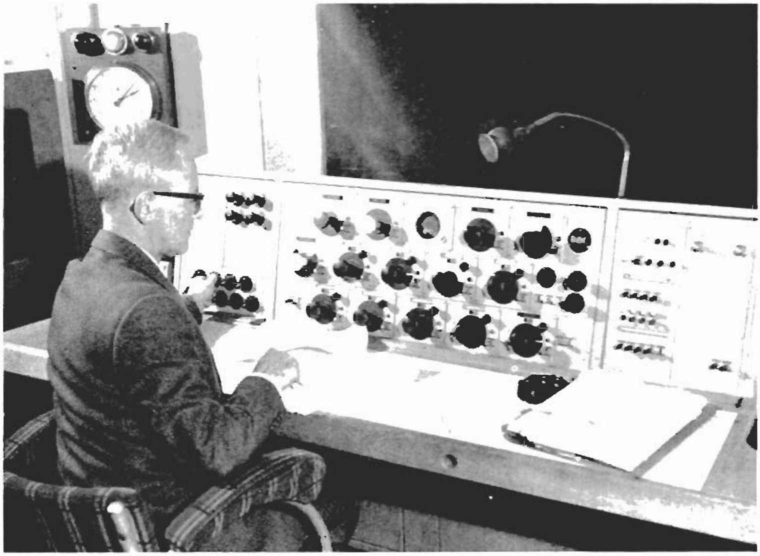


Plate 7.5. General purpose desk—Type B equipment

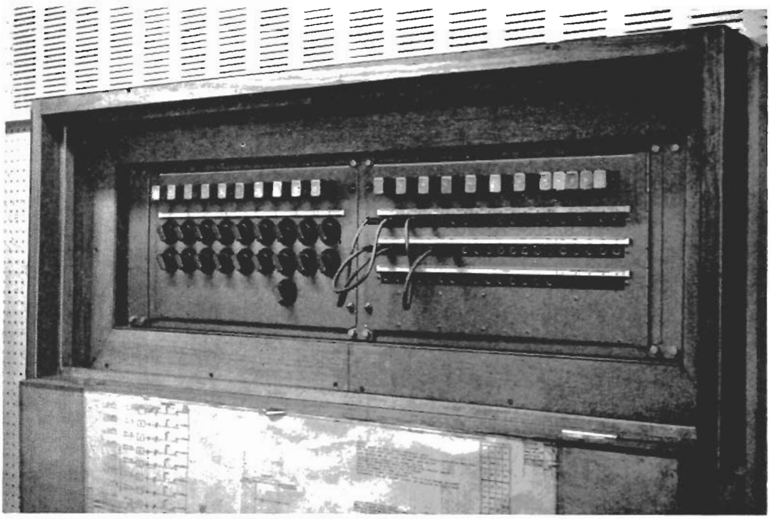


Plate 7.6. Type B equipment—pre-selection cabinet, showing cue-light switches on the left, and source/channel jacks on the right

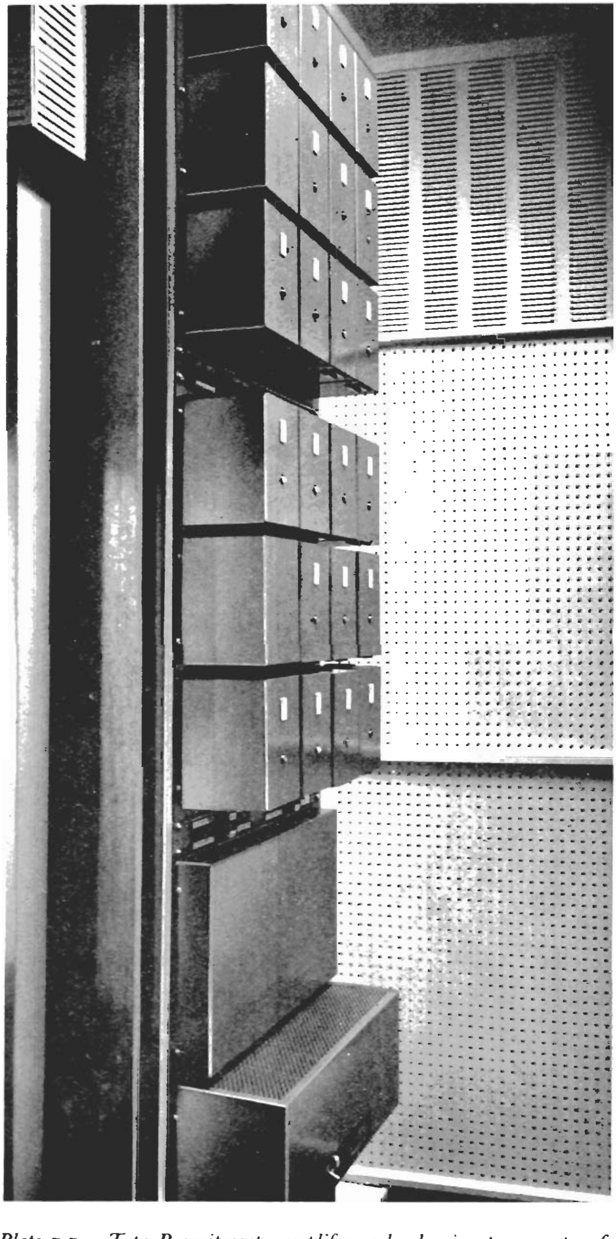


Plate 7.7. Type B equipment—amplifier rack, showing two groups of 10 amplifiers, standby amplifier and changeover switch, and mains unit



Plate 7.8. Control desk in Mixer Suite B.1., Broadcasting House



Plate 7.9. Special desk for BBC studio in Paris



Plate 7.10. Special desk for Covent Garden Opera House

plugged to any channel as required. Note also that material on any control room circuit not booked as a source to the mixer may be recorded in the mixer room by requesting that it be fed to the " miscellaneous record " jack.

7.6.4. Outside Sources

On the right-hand panel are situated 28 individual talk-back/pre-fade keys, which provide comprehensive facilities for passing directions to individual contributors. The keys are arranged in groups, and coloured to correspond with the arrangement of channels on the centre panel.

In the *normal* position of each key, cue programme is fed to the source, there being a *cue* programme selector which, on point 11, feeds the mixer desk output, and a change-over key corresponding to the " white " key alongside the mixer programme selector. Note that in the case of the *independent* channels W, X, Y, and Z, this cue programme will normally take the form of a *clean feed* on point 11.

In the *up* position of each key (locking) *pre-fade* listening on the associated source is provided on the desk headphones—and also on the mixer room loudspeaker during rehearsal or recording.

In the *down* position of each key (non-locking), the cue feed to the particular source is replaced by talk-back, and the desk headphones are switched to pre-fade listening. During rehearsal, the mixer room loudspeaker is also switched to pre-fade in this condition, but at reduced volume to avoid howl-round. During transsmision, pre-fade may be switched to the L/S by the transmission Pre-fade Listen switch. A special white key on the left-hand panel enables the studio manager to switch his headphones to *programme*, even when the producer may be pre-fading on the loudspeaker.

Only one of the individual talk-back/pre-fade keys should be used at a time. Two " master " talk-back keys on the centre panel give simultaneous talk-back to all sources on the two group faders, but the only means of speaking down the clean feeds to sources W, X, Y, and Z is via the individual keys. In a case where only clean feed is required without inadvertent use of talk-back—the genuine " multi-way " programme—this must be booked specially and Control Room will render clean-feed talk-back inoperative on transmission.

To avoid the automatic dimming of the mixer room loudspeaker, which accompanies use of the ordinary talk-back microphone, an auxiliary socket is fitted on the skirting-board for connection of a

lip microphone. When this connection is made, the lip microphone replaces the ordinary talk-back microphone, and the loudspeaker is not dimmed when talk-back is in use.

7.6.5. Green Cue Lights

There is a green key to the right of each of the Independent channel faders. Operation of these keys energises the " Master Cue " relays of any studio in Broadcasting House which is plugged as a source on the corresponding channel. In addition, a green key alongside each group fader will operate the master cue circuit of any Broadcasting House studio on that group whose channel is *faded up*.

7.6.6. Studio Red Lights

When the red light is applied to the mixer for transmission or recordings, the red light is also operated for any studio in Broadcasting House, Maida Vale, or 1 & 1A Portland Place, which is *connected* to a channel fader. In the case of four-source faders, the studio red light is applied when the corresponding indicator lamp is lit. This means that deliberate selection of the source will apply the remote red light when required—either for cueing a studio or for starting a reproduction from any tape reproducing channel which has been selected as a source on the mixer control desk.

7.6.7. Clean-feed Facilities

As mentioned above, the normal feed of cue programme to sources W, X, Y, and Z takes the form of a clean feed. Thus, with the special *cue* programme selector switch at point 11 (or the associated change-over key *down*), each outside source receives a clean feed of any of the other 27 sources which are faded up—on the control line for O.B.s, etc., and on line 10 for local studios.

7.6.8. Communications with Engineer's Cubicle

A talk-back unit providing two-way communication with the engineer's cubicle is situated on the right-hand panel of the desk.

7.7. DESKS FOR PARTICULAR APPLICATIONS

In addition to the special versions of standard studio desks described above, it is necessary, from time to time, to produce specially designed desks to fulfil a particular function. Two examples will be described.

7.7.1. BBC Studio in Paris

Plate 7.9 shows the desk designed for the BBC studio in Paris. Since most of the work of this studio is transmitted over land line to London, special talk-back and cueing arrangements are included. The two tape machines inset into the left-hand " wing " of the desk can be remotely controlled from the desk itself. Replay, record, and fast spool facilities are provided in this way.

The desk provides means of controlling a second, smaller studio in addition to the main one when, for instance, a narrator is necessary for a feature programme.

7.7.2. BBC Equipment at Covent Garden Opera House (Plate 7.10)

The listening room at Covent Garden is high up at the back of the gallery, and the acute viewing angle to the stage made a " flat " design essential. Quadrant faders were used, and various facilities were provided. In addition to the main slung orchestral microphone and two stage microphones mounted in the " float " there is provision for orchestral microphones in the pit, and microphones for " off stage " effects. A channel for artificial reverberation is provided. Two independent channels for narration and announcements are wired to extensions in one of the theatre boxes. Talk-back and cueing to the announcers and stage are available.

A somewhat unusual facility is provided in that it is possible to obtain two feeds to line, one the normal domestic feed including local announcements, the other being music only, for transmission to, say, a foreign radio organisation who can then add their own announcements. This latter is also termed a " clean feed ", a slightly different use of the term to that mentioned earlier.

7.8. MODIFICATIONS TO FREQUENCY RESPONSE

For all normal applications, studio and transmission equipment is designed so that it has a flat frequency response within the audible spectrum. There are occasions, however, when it may be necessary to introduce variations in this response, either for deliberate special effects, or to correct acoustical or recording quality. Some examples of this will now be given.

7.8.1. Microphone Correction Units

These are primarily designed to offset the increase in low frequency response that takes place on close speech—particularly

with ribbon microphones. (See Chapter 5.) The circuit and frequency characteristics of a typical M.C.U. are shown in Fig. 4.14.

A close technique, and therefore a Bass Correction Unit will be necessary in the following circumstances:—

(a) In small rectangular studios boomy quality is often caused by dimensional resonances, despite extensive acoustic treatment. Introducing some degree of bass correction permits a closer technique, and at the same time reduces the studio coloration—which is often at low frequencies.

(b) In large orchestral studios, a normal working distance for announcements may give unduly reverberant speech. Occasionally, therefore, the announcements may be made from less than 2 ft away, and bass correction used. It is desirable, at the same time, to ensure that the announcer sounds " in the same place " as the orchestra, etc.—i.e. in the same " acoustic ".

(c) For dance band vocalists, a very close technique has become the rule, and a number of portable bass correction units have been made available for London studios. They are labelled:—

" Bass Cut for Ribbon Microphone only at less than two feet ".

7.8.2. Portable Effects Unit PEU/1

This unit provides five amounts of Top Cut and four of Bass Cut, selected by separate switches. As might be expected, the overall

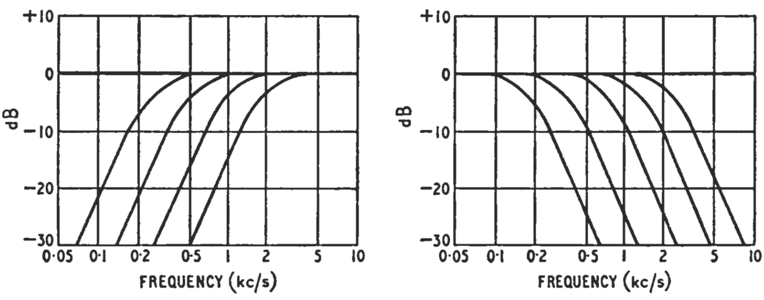


Fig. 7.26. Response curves of PEU/1

level depends to some extent on the degree of cut, and a careful note of the control settings should be made at rehearsal.

The PEU/1 is recommended when fixed amounts of distortion are required for special effects. It is much to be preferred to

fastening a baffle on to the microphone. The latter procedure may succeed for one voice in a fixed position, but gives uneven and unpredictable results in more complicated layouts, or where movements take place. It can never be used for serious music.

An idea of the distortion to expect from the PEU/1 may be obtained from the response curves shown in Fig. 7.26.

7.8.3. Variable Correction Unit VCU/1A

It is occasionally desired to transmit programmes which have been recorded with a different characteristic from that used by the BBC. Accordingly, the VCU/1A is fitted in certain channels, and may be adjusted aurally for best transmitted quality. For example, a Top Cut at 6 kc/s might be considered desirable on an overseas despatch, or on a Home programme which contained a number of very old disc inserts.

There are four correction circuits which may be used independently or in any combination:—

1. Bass Increase-decrease— } to vary the general shape of the
2. Top Increase-decrease— } response curves
3. Top Cut—to suppress surface noise etc. (4, 5, 6, 7, or 8 kc/s)
4. Bass Dip—to suppress mains hum (50, 60, 100, or 120 c/s)

Occasional use of VCU/1A in studios has been made to eliminate variations in the recording characteristics of some commercial microgroove recordings.

7.8.4. G12 Filter—Fitted on Turntable Desks TD/7

The BBC uses a different frequency characteristic for 78 r.p.m. discs than do commercial companies. For this reason, and so as to take full advantage of the relative freedom from surface noise which is possible in direct recordings, turntable desks include a two-position filter switch. This allows both types of disc to be reproduced, the overall levels being matched up ($\pm$ 4 dB) at the same time. The effect on a direct recording of the wrong key setting is loss of top; and for a gramophone record produces excessive surface noise.

The materials used in recent BBC pressings can result in a freedom from surface noise which is comparable with that of a direct recording. New pressings—especially pressings of music—should therefore be played in the " Direct " position unless surface noise is particularly noticeable.

7.8.5. "Quad" Quality Control Unit

This is a commercially manufactured domestic pre-amplifier/tone control unit, which has been modified for use in BBC studios. As modified, it has zero dB gain, and 600 ohm input and output impedances. The volume control is rendered inoperative.

The first BBC studio application of this instrument was in gramophone studios, where it was arranged so as to be available on

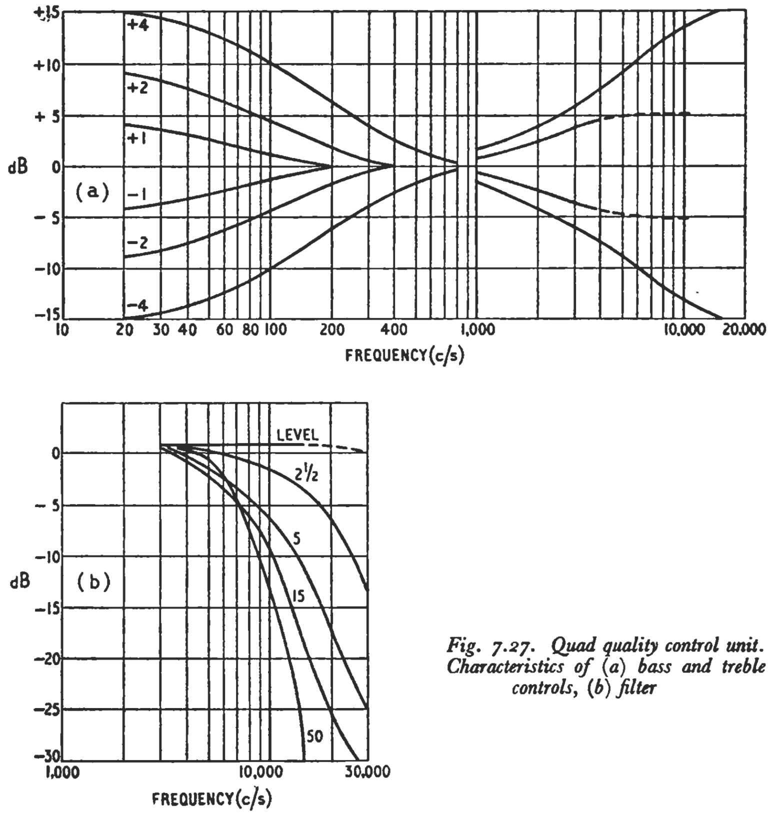


Fig. 7.27. Quad quality control unit. Characteristics of (a) bass and treble controls, (b) filter

TD/7 outputs only. Its purpose was then to apply filtering to 78 r.p.m. records to remove surface noise, excessive high frequency emphasis, and distortion. A key was provided to cut the apparatus in and out of circuit. With the advent of the RP2/1, these units have now been connected in such a way that they can be used on all

types of gram desk, and so obviously become available for use on fine-groove discs.

It must be stressed that the use of the Quad should be confined to discs where there are faulty conditions to correct. Any attempt by a broadcasting organisation to alter the balance or general sound of a commercial disc is liable to incur the displeasure of the manufacturers!

More recently, these units have been used experimentally in microphone circuits, mainly in light music and light entertainment programmes, both in order to achieve better separation between sections in a multi-microphone balance, and for special musical effects.

The facilities provided are as follows. Bass lift and cut at a rate of 6 dB/octave, with frequency of turnover variable, such that at maximum boost, the turnover is approximately 1 kc/s, with a maximum variation of $\pm$ 10 dB at 100 c/s. Treble lift and cut, turning over at 1 kc/s, with a maximum of $\pm$ 15 dB at 10,000 c/s. It must be noted that the calibration of $-$ 4, 0, $+$ 4 on these two controls is purely arbitrary, and not a dB scale (Fig. 7.27 (a).

A low pass filter is also available, having a choice of three turnover frequencies, 5, 7, and 10 kc/s, the slope of the roll-off being continuously variable between 0 and 50 dB/octave. In this case the control is actually calibrated in dB/octave (Fig. 7.27 (b).

The push-buttons on the unit are normally inoperative for BBC use, but an experimental version has been produced where these are used to switch into circuit either of two presence filters, giving 2, 4, or 6 dB rise at 2·8 and 6 kc/s.

8

OUTSIDE BROADCASTS

8.1. OUTSIDE BROADCASTING EQUIPMENT

A temporary rig capable of easy dismantling, yet providing elaborate facilities when required, is the general description of a set of O.B. equipment. Only the basic layout for the OBA/8 and the OBA/9 equipments will be described. The more complicated programmes call for an extension of these layouts in an almost infinite variety of combinations.

8.1.1. OBA/8 Equipment

This equipment consists of the mixer MX/18, the amplifier OBA/8, and its associated power supply, together with a monitoring loudspeaker and amplifier (Plate 8.1 and Fig. 8.1).

(*a*) *Mixer MX*/18 (Fig. 7.5.)

Each channel of this mixer is a balanced potentiometer introducing attenuation in 2 dB steps, with 3 dB increments on the last four studs. A study of the diagram shows that each channel " sees " the output with the other three channels in parallel. When the other three channels are faded out, Microphone 1, for example, is correctly matched into the amplifier (300 ohms). But fading up a second channel (with microphone or grams attached) upsets the matching, and a drop in level of as much as 3·5 dB occurs—an amount which may call for compensation on the main control potentiometer.

Fading up further channels results in more loss, in smaller increments. Three extra channels, for example, reduce the level by about 8 dB.

In the diagram a high resistance or " static leak " is connected from the end of the resistor to the *off* stud in order to preserve electrical continuity and prevent clicks due to building up of

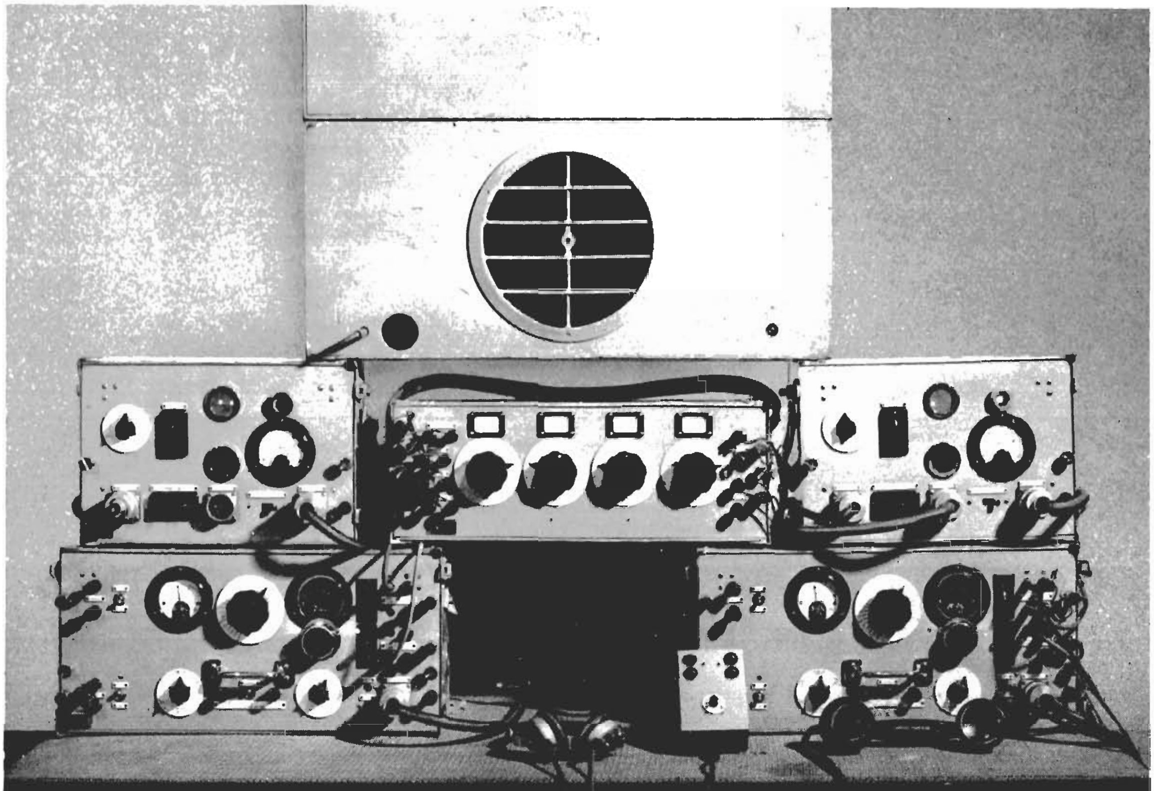


Plate 8.1. Typical arrangement of OBA/8 equipment

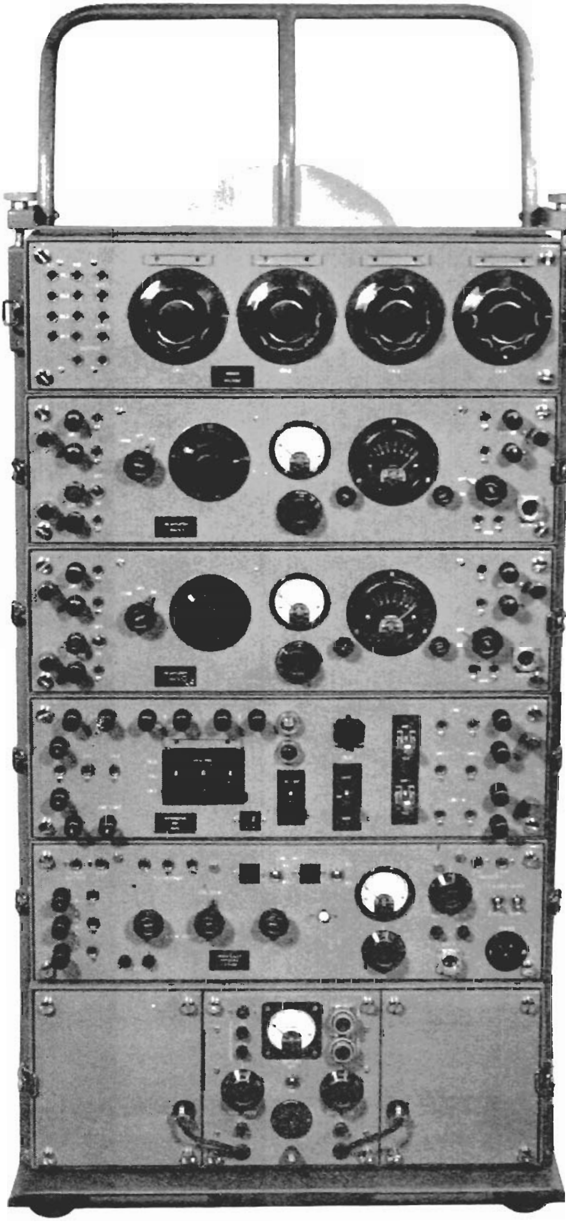


Plate 8.2. OBA/9 trolley



Plate 8.3. Mobile Control Room



Plate 8.4. Self-operated O.B. equipment

charges on the contacts. Only one of these is shown for simplicity, though more may be used.

(*b*) *Outside Broadcast Amplifier OBA/8*

This is a two-stage amplifier, incorporating a gain control and the necessary valve circuits to feed a peak programme meter. The input impedance of the amplifier is 300 ohms, and the maximum gain is 91 dB (control at 35). Fading-down introduces attentuation at

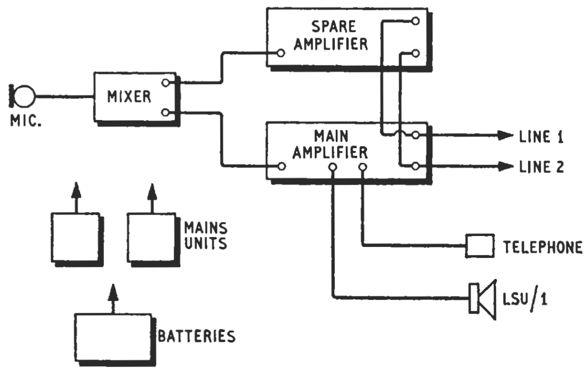


Fig. 8.1. Block diagram of O.B. rig and line connections

2 dB per stop from 35 to 11, and 3 dB from 11 to 1, with fade-out at zero. At an average setting of 25, the gain is about 70 dB, thus raising the output of microphones or grams to about zero level.

An *output attentuator* introduces 0, 4, or 8 dB attenuation, giving output levels (P.P.M. reading 4) of + 4, 0, or − 4 dB respectively. The output circuit is loaded in the receiving control room in such a way (239 ohms) as to make this sending level correspond to *zero voltage* level. (For definition see Appendix.)

(*c*) *Calibration of Programme Meter*

Regular calibration is necessary and in order to do this the following steps should be taken:—

(1) Allow 10 minutes for amplifier to warm up, after switching on.
(2) Fade out main control.
(3) Unplug loudspeaker if LSM jack is used.
(4) Adjust zero by means of the control marked " Adj. zero ".

(5) Throw the calibration switch to CAL.

(6) Adjust meter to read 4 by means of the control marked " Adj. Sensitivity ".

(7) Return switch to Normal and re-plug the loudspeaker.

For O.B.s, a minimum of two amplifiers and mains units are installed as a precaution against breakdown, and two sets of batteries are also connected in case of failure of the mains supply. Provision is made for rapid interchange of the programme and control lines to the studio centre.

The block diagram shows the mixer MX/18 connected to the amplifiers in parallel. The input switch of the spare amplifier is at *off*. The output jacks of the amplifiers are also paralleled, using double ended cords. The normal condition is main amplifier output to Line 1, telephone to Line 2, this being arranged by suitable positioning of the output keys.

(*d*) *Amplifier Change-over*

In the event of failure of the main amplifier, or its mains unit, the input switch on the spare is thrown to *on* (the gain control being at a suitable setting) and the other switched *off*. It is only necessary then to transfer the loudspeaker cord to the LSM jack on the spare amplifier.

During this change-over, the amplifiers are operating in parallel for a second or so. It is therefore usual to test during the installation that the feeds from the mixer are " in phase ", otherwise cancellation would take place in the output to line.

(*e*) *Line Change-over*

If the programme line—Line 1—develops a fault, the control room will telephone on Line 2 requesting a line change-over. After a pre-arranged interval—say, 15 seconds—the Line 2 change-over key should be thrown from TELE to AMP, and after a pause, Line 1 key may be moved from AMP to TELE. Exactly on the 15-second cue, the control room will have changed over the termination of Lines 1 and 2 by re-plugging, and will then telephone to confirm satisfactory results.

8.1.2. OBA/9 Equipment (Plate 8.2)

A considerable amount of auxiliary equipment is necessary with the OBA/8 rig, and the OBA/9 was developed in 1950 to simplify transportation, etc.

The equipment fits on to a trolley, which can be carried in a large car. The illustration shows the front of the trolley, with mixer MX/29, spare and main OBA/9, distribution unit, LSM/9 and supply unit. On the back of the trolley are three drums of microphone cable which can be run out without dismantling.

The *mixer unit MX/29* provides for four low level input channels of 300 ohms or 30 ohms impedance. Attenuators are also included in case one high level source is to be mixed.

The *amplifier OBA/9* provides similar facilities to the OBA/8, except for output line switching.

The *distribution unit DU/1* provides flexible connection of programme and telephone outputs to the studio centre, as well as to on-site commentators, etc.

The *LSM/9 unit* feeds the monitoring loudspeakers, and also supplies trap-valve feeds to public address systems, and other destinations as required.

The *supply unit SUP/6* contains both batteries and a mains unit. The normal transmission arrangement is main amplifier mains operated, and spare amplifier connected to the batteries.

8.2. O.B. LINES

8.2.1. Termination of Lines at O.B. Point

At *permanent* O.B. points, the microphone extension terminals and the P.O. Exchange lines should all be clearly numbered and labelled. The numbering of the P.O. lines should correspond with

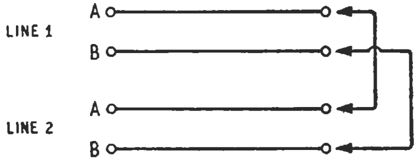


Fig. 8.2. Looping of lines

the local end numbering in the BBC control room through which the O.B. is routed.

At *temporary* O.B. points, the P.O. line terminations should be clearly labelled as soon as they can be checked over with the control room concerned.

When an O.B. point is left unattended, it is standard practice to leave the lines *looped* together in a certain way, to enable conclusive tests to be carried out from the BBC control room or P.O.

exchange. The method of looping is A-A and B-B, as shown in Fig. 8.2.

When an odd number of circuits exists—say, one music and control plus a cue or spare music line—the odd line is looped A-B. When an O.B. point has its exchange lines connected through to a BBC control room, the lines must be left looped in accordance with the above if the engineers are absent from the O.B. point for periods exceeding half-an-hour.

8.2.2. Tests on Lines from an O.B. Point

It is the duty of the senior O.B. engineer to ensure as soon as possible after arrival at an O.B. point that tests are carried out on the exchange lines.

The lines may be connected direct to the local BBC control room (see Fig. 8.3 (a)), or to the P.O. exchange for extension to trunks or other circuits on to BBC networks (see Fig. 8.3 (b)).

The BBC control room in (a) will probably have connected a telephone ringing indicator to one of the circuits in anticipation of

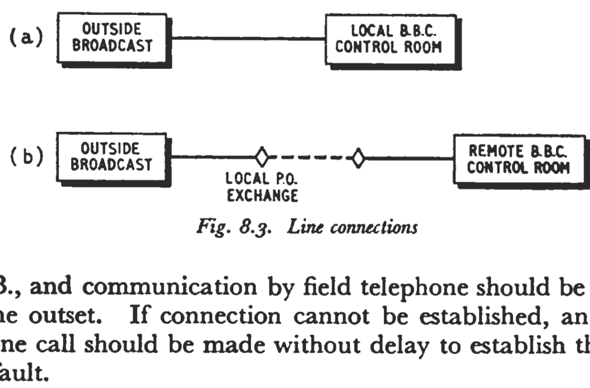


Fig. 8.3. Line connections

the O.B., and communication by field telephone should be possible from the outset. If connection cannot be established, an outside telephone call should be made without delay to establish the cause of the fault.

In (b) an ordinary telephone call to the local P.O. exchange should be made to arrange the tests.

The tests consist of speaking on each circuit in turn, and co-operation in D.C. tests as required. In the D.C. tests, a *short circuit* is applied to each line in turn, to allow the remote engineer to measure the resistance of the line. Then the ends are left open while insulation between the two " legs " and earth are tested.

A *programme test* should also be originated from all O.B. points, to enable the control room to judge the incoming quality. Part of

the rehearsal may be used, or if this is not possible, speech on one of the microphones installed for the O.B., or special one installed for the purpose near the apparatus. This programme test is particularly important where the control circuit is unsuitable for music, as it will disclose cases where the music and control circuits have been inadvertently crossed.

8.3. CO-OPERATION BETWEEN O.B. POINT AND CONTROL ROOM

It will be appreciated that during the final period before the O.B. starts, the fullest co-operation must be extended by both ends. This will consist in general of answering the telephone promptly, or warning the other end if for any reason this will not be possible during any particular period.

8.4. SELF-OPERATED O.B. EQUIPMENT

This is a small, self-contained equipment for use by commentators on occasions where there is no need for a full scale outside broadcast rig, and where no engineer is necessary (Plate 8.4).

The equipment consists of a suitcase-mounted amplifier with line switching equipment to enable the necessary pre-transmission tests to be made. Talk-back and cueing from the receiving control room or studio are possible on headphones, and a miniature receiver is provided for transmission cue.

The amplifier is battery operated, transistorised, and the microphone used is the Acos Mic.39.

9

PUBLIC ADDRESS SYSTEMS

9.1. INTRODUCTION

In studios where an audience is admitted, and more particularly in studios where Light Entertainment programmes are taking place, it is necessary, in order that the audience may hear the broadcast more satisfactorily, that a public address system should be installed. P.A. equipment is also used under Outside Broadcast conditions, when if an existing installation is present this must be linked to the broadcasting equipment, and if not a complete system must be specially rigged for the occasion. The use of public address introduces a number of problems and in this chapter some of the difficulties will be described.

In any P.A. system the basic requirements are obviously a microphone, a power amplifier and one or more loudspeakers. The microphone may be a separate instrument or, and this is more likely, be one or more of the microphones used for broadcasting purposes, the output of these microphones being split so that they can feed both the broadcast and public address systems.

The effectiveness of a P.A. installation may be judged under the following headings:—

(a) Loudness
(b) Quality
(c) Instability
(d) Sense of direction

9.1.1. Loudness

Correct loudness at all points in the auditorium is not easy to achieve. If there is sufficient volume of sound to satisfy the back rows of seats, there is a danger of excessive loudness for people near the loudspeakers. The cure for this is to mount the loudspeakers

high, so that the nearby audience is on the fringe of the loudspeaker's polar diagram. Since these people are near the stage, direct sound will ensure that they hear well. The back rows of seats receive a useful reinforcement of sound by reflection from the rear wall (see arrow in Fig. 9.1), but angling of the loudspeakers

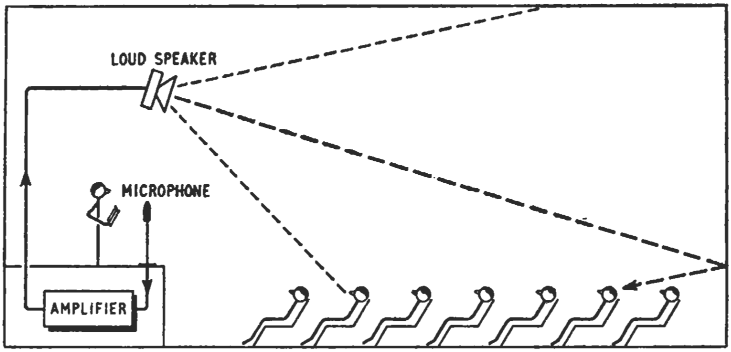


Fig. 9.1. Basic Public Address system

must be such as to prevent these reflections reaching seats further than 20 ft from the rear wall, or else *echoes* will be introduced.

9.1.2. Quality

The quality of the reproduction is at best a compromise, bearing in mind the varying directivity of loudspeakers, and the overlapping areas covered by the direct sound and the loudspeakers. For best intelligibility there should be some attenuation of the bass frequencies, and this can be done in the P.A. chain, either by using loudspeakers with inefficient bass response or by means of deliberate filtering.

It follows that deterioration of broadcast quality will result if much of the P.A. radiation is allowed to spill on to the microphones—either stage microphones, or those used to pick up audience reaction—and the siting of microphones and loudspeakers must be regarded as an inter-dependent operation.

Correct *phasing* of the loudspeakers is important. To check this, it is usual to match up the volume from each loudspeaker individually, listening at some point equidistant from them, and then to switch in both together and arrange the terminal connections for

an improvement in volume. This test is better carried out on low frequency tone, or even mains hum.

9.1.3. Instability

Instability may occur when the microphone feeding P.A. is itself within range of the loudspeakers. There is, in such circumstances, a danger of the sounds being repeatedly amplified—until a howl-round takes place—causing loud oscillations if it is permitted to build up. Incipient cases may not bring about continuous howl-round, but the resulting distortion is enough to mar programme. The solution again lies in correct siting of the microphones and loudspeakers, and correct setting of the P.A. gain control.

The relative phase of the microphone to the loudspeakers will often affect the onset of howl-round. This may be checked by fading-up to the point of instability, and reversing the microphone

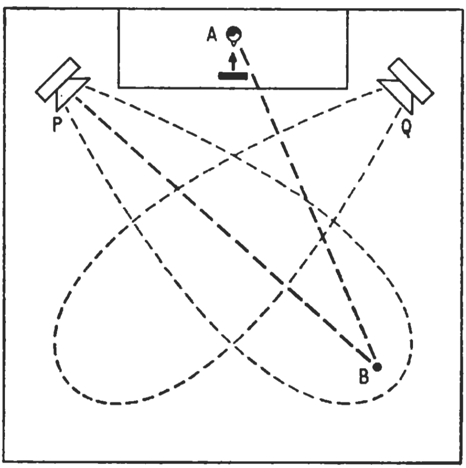


Fig. 9.2. Siting of loudspeakers to ensure good "sense of direction"

terminals—or the microphone itself in the case of ribbons—to discover the safer condition. When roving microphones are used, the problem is made more difficult.

9.1.4. Sense of Direction

Other things being equal, our sense of direction, when listening to more than one source of a given programme, works in such a way that we associate direction with the source which *reaches us first.* Within certain limits, this remains true even if the later sounds are stronger than the first. In P.A. work it is desirable, therefore, to

arrange that the path from loudspeakers to the audience shall always exceed the " direct " path. In Fig. 9.2, for example, two loudspeakers, P and Q, are directed at opposite diagonal corners. PB exceeds AB, and provided loudspeaker Q is angled sufficiently to



Fig. 9.3. Typical polar diagram of line source at high frequencies

miss the listener at B, the sounds will appear to come from the actor at A. The worst conditions exist for people near the two loudspeakers, but some improvement is often possible if the loudspeakers are placed high (as recommended in Section 9.1.1. above), so that the sound heard is almost entirely " live ".

While stressing that the reinforcement path should not exceed the direct path, it is necessary to say that they should not differ by more than about 20 ft, since beyond this distance a discrete echo becomes apparent.

9.2. TYPES OF LOUDSPEAKER

The loudspeakers commonly used for P.A. can be divided into three groups, arranged in increasing order of directivity, as follows:

(a) ordinary cone loudspeakers

(b) horn loudspeakers

(c) line source loudspeakers

A single *horn loudspeaker* is more effective than a cone loudspeaker, since its narrower polar diagram permits more accurate direction of the sounds at the audience, with consequently less spill on to microphones. Furthermore, the falling off of intensity along the sound path is less steep, giving more even loudness to near and distant listeners.

The *line source*, or " column ", loudspeaker carries the " focusing " process a stage further. It comprises an array of loudspeakers mounted in line in a unit 6 ft or more in length. The radiation of sounds—particularly at high and middle frequencies—is directed along the plane at right angles to the column, in a fan-like manner (Fig. 9.3). If the unit is set up vertically, as is most usual, unwanted

and wasteful radiation of sounds upwards, or downwards on to the heads of nearby listeners, is reduced—with a desirable saving in the electrical power necessary to cover a given auditorium. It is important to note, however, that no particular directional effect is achieved in the horizontal plane in this case, and if this is required, the column should be tilted, or even set up horizontally.

9.3. FREQUENCY SHIFT P.A. SYSTEM

A recent development in public address equipment has been the result of some research in the United States, and this equipment has now been used with considerable success in some BBC studios.

In this equipment an electronic device is inserted between the microphone and the loudspeaker equipment, and this device shifts all frequencies passing through the system by a small amount, usually of the order of 5 c/s. The object of this is that at any given instant the direct sound arriving at the P.A. microphones will be at a different frequency (by 5 c/s), from the sound arriving at the P.A. loudspeakers, so there is less likelihood that the system will howl. In practice a howl point is eventually reached, but an improvement in P.A. loudness of upwards of 6 dB is normally obtained. In conditions of high ambient noise this improvement may not be apparently as great as this, since the background noise is also amplified by the P.A. system and so the ratio of programme to background noise may not be significantly improved. This, of course, happens with ordinary P.A. systems but the added gain with frequency shift may make the effect more noticeable.

9.4. DELAYED SYSTEMS

If public address is to be achieved in very large halls, or in the open air, members of the audience may be at a great distance from the original sound source, and it will be necessary to place a series of loudspeakers at various points between the back row of the audience and the sound source. If these loudspeakers are connected normally to a P.A. system, the time taken for the sound to reach the audience from the original sound source will be much longer than that from the nearest loudspeaker, and so the sound will appear to come not from the original source, but from the loudspeaker itself. By a system of progressive delay from the original source to the most distant loudspeaker this effect can be minimised in order to preserve the condition that the original sound source must be heard first, if only rather weakly.

9.5. STEREOPHONIC PUBLIC ADDRESS EQUIPMENT

It may be required, under some circumstances, to reproduce stereophonic recordings introduced by speech from a stage, to a large audience. This is not normally successful if only one pair of loudspeakers is used. The situation can be improved, and a good stereophonic effect achieved over the greater part of the audience area, if a number of pairs of loudspeakers, arranged down the

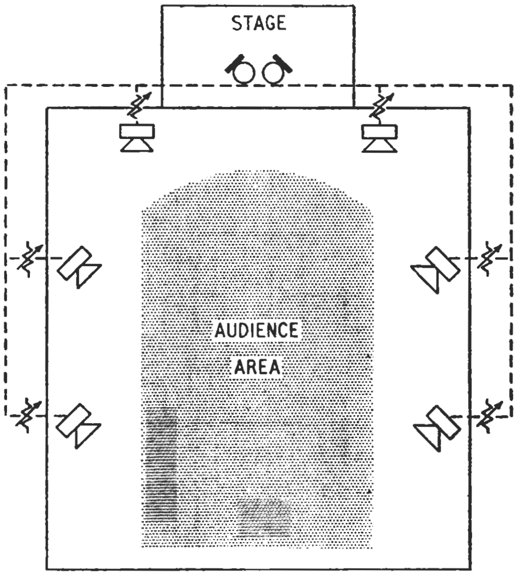


Fig. 9.4. Stereophonic P.A. installation

sides of the hall are used. These loudspeakers must all be in phase with one another and each pair must be correctly balanced in order that a monophonic signal, fed equally into the two halves of the system, shall produce a centre image. Once each pair has been balanced in this way it is necessary to adjust the relative volume levels of each pair of loudspeakers with respect to the preceding and following pairs, such that the audience at any given point in the sitting area will only hear the pair of loudspeakers which are nearest to them (Fig. 9.4).

Announcements can be fed into this system in two ways. The simplest form, if only one person is speaking, is to use a monophonic microphone and to connect its output equally to the two

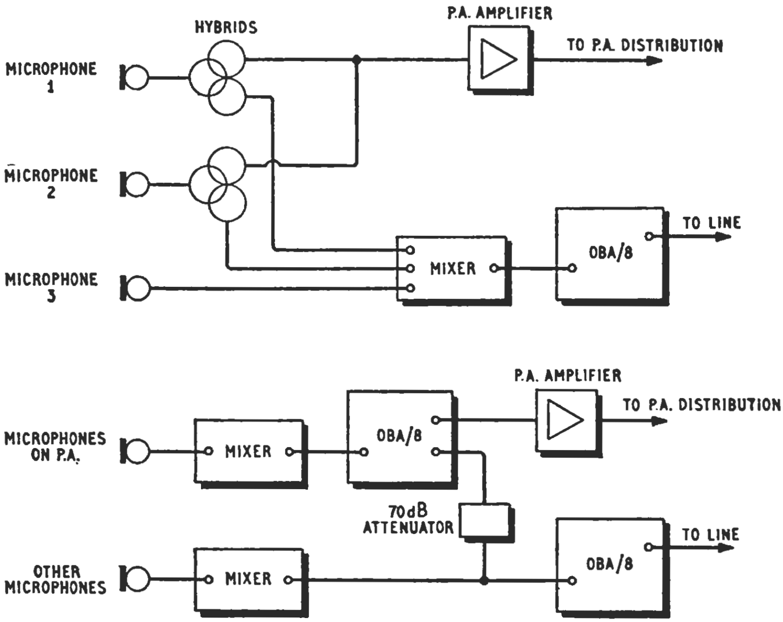


Fig. 9.5. P.A. in OBA/8 rig

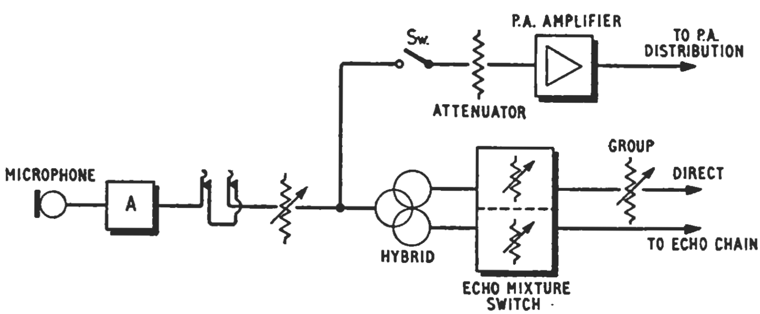


Fig. 9.6. P.A. in Type A Equipment

stereophonic channels. Alternatively, if two or more persons are speaking it may be possible, under some acoustic conditions, to achieve a fair stereophonic effect by using a stereo P.A. microphone.

The success of the reproduction of stereophonic material in this way cannot be predicted, but is more likely if the general acoustic is dead. However, moderate success has been achieved in a number of halls having widely different acoustic conditions.

9.6. P.A. CIRCUITS

The use of hybrid transformers to derive a P.A. feed from each programme channel is also described in Chapter 7.

In *OBA*/8 installations, there is no special provision for P.A., and separate hybrid transformers are simply wired at low level at the input to the mixer, or sometimes between the mixer and the OBA/8. Fig. 9.5 shows an alternative arrangement using a 70 dB attenuator, and no hybrids.

With the *OBA*/9, the same methods are used, with the addition that, in cases when the complete programme is to be fed to P.A., the trap-valve amplifier supplies a feed at about zero level for the purpose.

With *Type A Equipment*, P.A. circuitry is included only in " audience " studios, and takes the form shown in Fig. 9.6.

The switches SW are supplied in a few studios such as the Concert Hall, and enable the studio manager to select sources to P.A. at will. In other studios, selection of channels to P.A. is made on the engineering control position, and often hybrid transformers are used in place of attenuators. In all Type A installations the P.A. level is dependent on the setting of both the channel fader and the volume control on the P.A. amplifier. Another point to notice is that fading out the group or main faders will not normally remove the feed to P.A. if the *channel* fader is open. In certain studios, however, complete fade out of group faders cuts the feed of that group to P.A.

With *Type B Equipment*, switches are fitted on the control desk for all channels which can be fed to P.A. The *up* position of these switches gives a 10 dB boost to the P.A. level from each channel. On Mark III desks there is an auxiliary fader giving overriding control of the volume of P.A.

It should be noted that on all the above circuits the various inputs to Public Address are derived from channels carrying contributions to the actual programme itself. However, this paragraph would be incomplete without mention of the simplest arrangement of all—namely, the use of a self-contained P.A. circuit. In this, a

separate P.A. microphone is set up alongside the broadcast microphone (or sometimes fastened to the same stand) and allows P.A. volume to remain quite independent of mixing levels on the broadcast programme.

9.7. P.A. EQUIPMENT IN CURRENT USE

A number of commercial loudspeaker amplifier combinations have been used by the BBC. At present the amplifiers most commonly employed for P.A. work are manufactured by Pamphonic Reproducers Limited. They have a maximum power output of 10, 25, or 50 watts, and will operate either from a balanced input at microphone level, or balanced or unbalanced inputs at zero level. They have output connections suitable for driving line source units or low impedance loudspeakers direct.

The loudspeakers most commonly employed are Pamphonic 6-ft or 8-ft line source units, using the principle outlined above. To give equivalent directivity at high and low frequencies, these units contain two lines of loudspeakers—bass and treble—with the necessary cross-over circuits. In studio installations it is usual to set up two columns to cover the floor area, and a further two for the gallery as necessary.

Other loudspeakers used in Outside Broadcasts, are 12-, 10- and 8-in. cone units mounted in box cabinets, the latter being strung out in relatively large numbers to give low volume coverage of audiences.

10

TAPE RECORDING AND REPRODUCTION

TAPE recording in its present form is a development from a system which grew up in Germany during the last war, but the principle is a relatively old one, indeed it was known at the time the first disc recordings were made at the turn of the century. In the early days, the recording medium was a thin steel tape and recording machines employing this were in use by the BBC until shortly after the war (Plate 10.1). This system suffered from a relatively poor frequency response, at least by modern standards, and a fairly high level of background noise. Other difficulties included the necessity for a welding technique to repair a broken tape or to make joints during editing.

The development of the "Magnetophon" tape recorder in Germany during the war, using a plastic tape coated or impregnated with magnetic material, made high quality recording possible, and the discovery that a high frequency bias applied during the recording process would keep the distortion to a very low figure paved the way to subsequent developments.

10.1. ELEMENTS OF A TAPE MACHINE

The tape recording and reproducing machine consists of a *tape transport system* which will move the tape at a constant linear speed across three tape heads. The first of these is the *erase head,* the function of which is to ensure that the tape is completely demagnetised. The second head is the *recording head,* which is fed with the programme currents which are to be recorded. The third head is the *reproducing head,* used to monitor the tape whilst recording and to reproduce the tape when required (Fig. 10.1).

The recording head is fed from a *record amplifier* which contains part of the compensation or *equalisation* necessary to produce an overall "flat" frequency characteristic. The reproducing head

feeds a *reproducing amplifier* and this contains the rest of the necessary equalisation. The erase head is fed at a high alternating frequency, usually between 60 kc/s and 100 kc/s, from the *bias oscillator*, and this oscillator, as its name implies, also feeds the high frequency bias to the record head (Fig. 10.2).

In the domestic tape recorder, one amplifier commonly takes on the function of either record or reproduce, the necessary change-over of equalisers and connection being made by the record/play

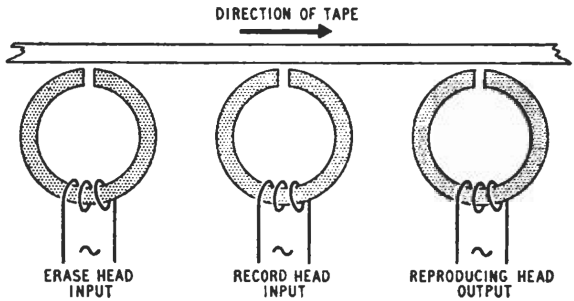


Fig. 10.1. Diagram of the three heads in a tape recorder

switch. Under these conditions a single record/play head is used, thus saving considerably on the cost of the equipment. Of course, under these conditions no monitoring from the tape during recording is possible and the recording cannot be heard until after it has ended, the tape rewound, and the equipment switched to play. Whilst this is perfectly satisfactory for many domestic applications, it is not practicable for a broadcasting system since it is imperative

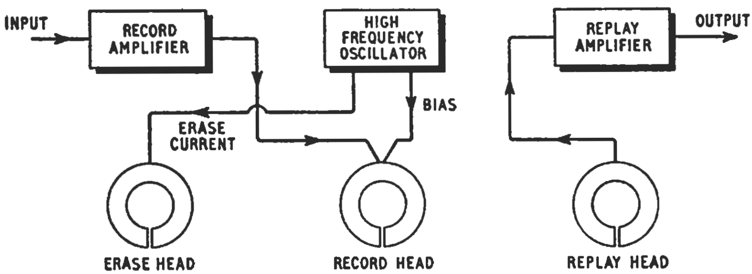


Fig. 10.2. General schematic of tape recorder

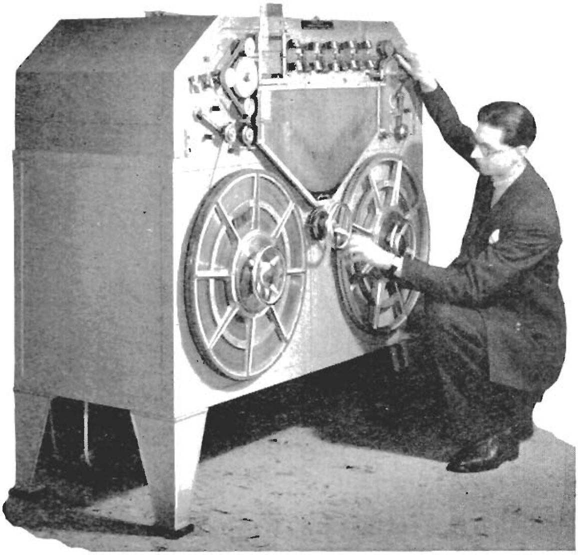


Plate 10.1. Marconi-Stille steel tape recorder

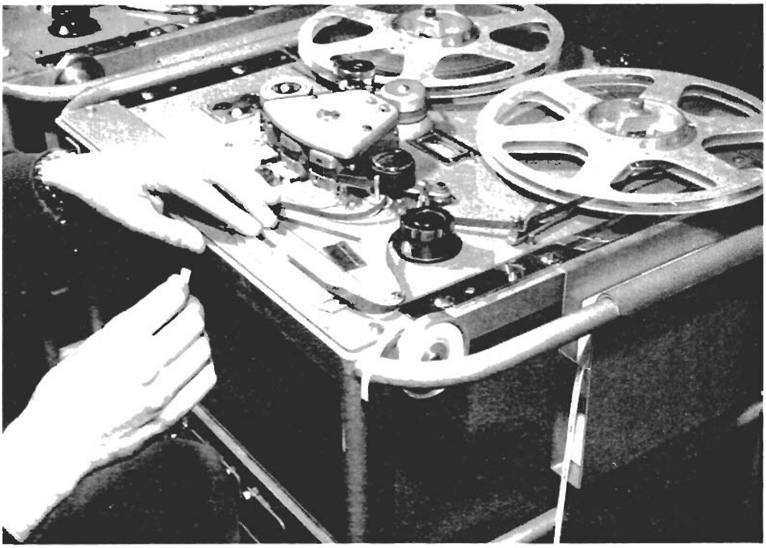


Plate 10.2. E.M.I. recorder TR/90 showing tape editing block



Plate 10.3. Ferrograph tape recorder



Plate 10.4. The Leevers-Rich tape recorder

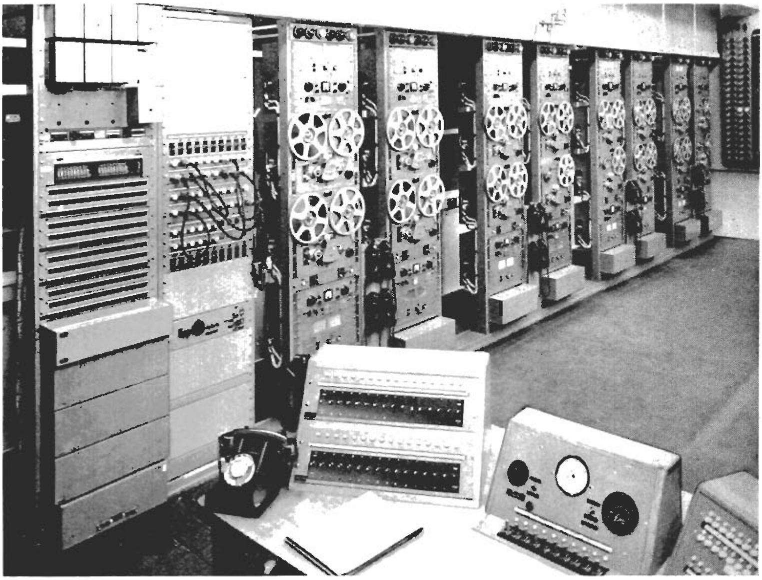


Plate 10.5. Central Recording Room H.18

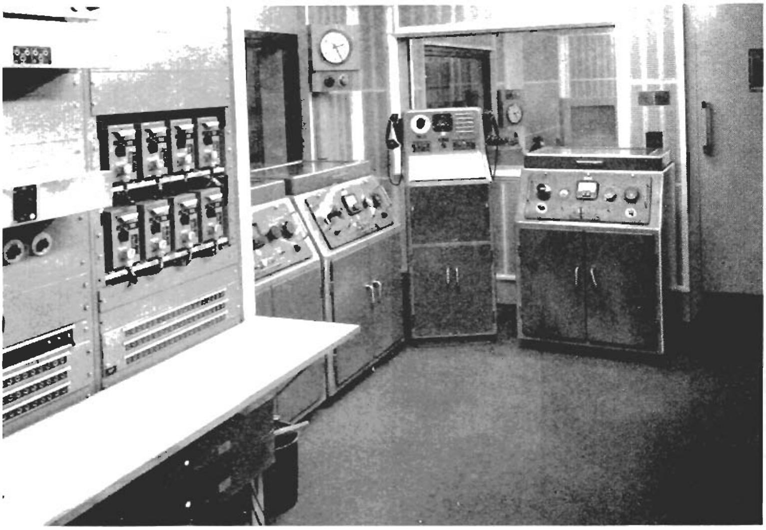


Plate 10.6. Engineer's control cubicle at the Paris studio in London, showing, in addition to the normal control bay, the three tape recorders for local recording. The Studio Manager's control cubicle can be seen through the far window

Plate 10.7. E.M.I. midget recorder





Plate 10.8. Ficord 1A recorder, showing extremely small size

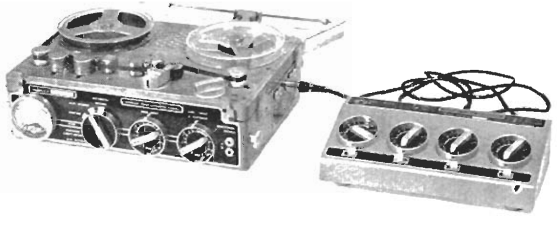


Plate 10.9. Nagra 111B recorder

to know that a recording is in fact on the tape, because of the high cost of repetition when expensive artists are involved.

10.2. OPERATION OF A TAPE MACHINE

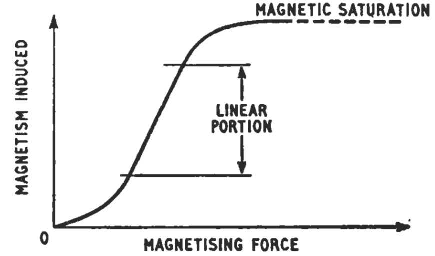
10.2.1. The Erasing Process

It is necessary before new material is recorded on a tape for the tape to be completely free from any previous magnetism. This is achieved by passing it across the erase head. This consists of a ring of laminations of magnetic material, having a small but finite gap (Fig. 10.1), the laminations being magnetised by an alternating current passing through a coil wound round them. This current at high frequency, as described above, will cause an alternating magnetic field to appear across the gap in the laminations. This field will " bow " out into the tape with the result that as the tape moves across the head each minute section of tape will pass first an increasing alternating magnetic field which will take it up to full magnetisation, and then immediately through a decreasing magnetic field which will reduce this magnetisation to zero.

10.2.2. The Recording Process

The completely demagnetised tape next passes across the recording head which is a similar ring of laminations, again having a

Fig. 10.3. Non-linearity of magnetisation characteristic



coil, through which, this time, the programme currents pass, having been amplified by the recording amplifier. These currents again produce a varying magnetic field across the gap in the laminations, and this field magnetises the tape in sympathy with the programme. In order for this to be achieved without distortion, it is necessary to apply a *bias* to the recording head at the same time as the programme. This is necessary because the relationship between the magnetising force and the magnetism induced in a magnetic

material, in this case the tape, is not linear—see Fig. 10.3. In the absence of bias, a sine wave input would produce a recorded waveform considerably distorted as shown. Fortunately the magnetisation curve has portions which are substantially linear and the bias assists in ensuring that the audio frequencies magnetise the tape

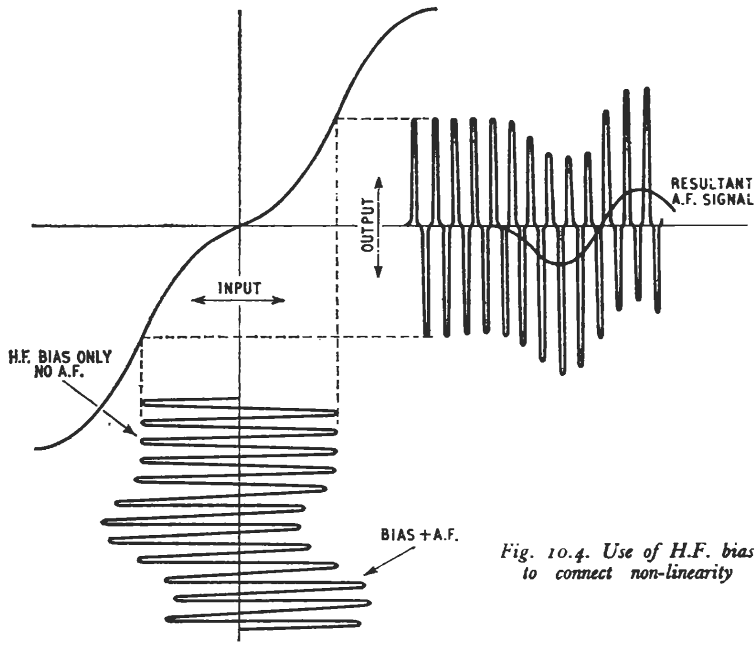


Fig. 10.4. Use of H.F. bias to connect non-linearity

on this linear part of its characteristic, as shown in Fig. 10.4.

In the early days of magnetic recording, a d.c. bias was used, and whilst this was successful in reducing distortion it resulted in a poor signal-to-noise ratio. The modern use of high frequency bias reduces the distortion and ensures that the signal-to-noise ratio has an acceptably high value.

10.2.3. The Reproducing Process

The reproducing head is again similar in construction to the recording head, and the changing magnetic state of the tape as it passes the gap in the head induces a changing magnetic field in the

laminations which, in its turn, induces an alternating current in the coil. This current is fed to the reproducing amplifier.

10.2.4. Equalisation in Tape Recording

When a recorded tape is passed over a replay head, the voltage produced at the terminals of the head will be equal to the number of turns on the core multiplied by the *rate of change* of flux in the core. The rate of change will depend upon the speed of the tape, and the *frequency* of the recorded material. If the recording has been made at constant r.m.s. intensity, the output voltage from the replay head will be doubled every time the recorded frequency is doubled; that is, it will rise with frequency at the rate of 6dB/octave (see Fig. 10.5). Clearly this is not a flat response, and some equalisation is necessary. Equalisation for this rising characteristic is contained in the replay amplifier.

At very low frequencies, and at very high frequencies, further difficulties occur.

At very low frequencies, due to the fact that the " magnets " recorded on the tape are longer than the length of tape in contact with the pole faces which contain the gap, not all the flux will close round the core of the head, and some of the path of the flux will be in the air. Because of this more losses occur, and the bass response falls at a rate considerably in excess of 6dB/octave.

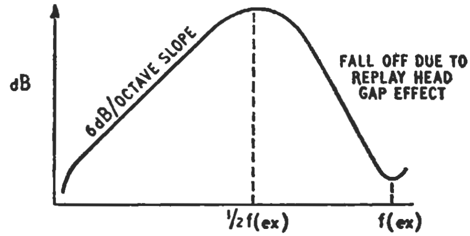


Fig. 10.5. Unequalised tape recording characteristic

Equalisation for this loss is usually only found in professional recording equipment, and is situated in the recording amplifier. At very high frequencies, as the recorded wavelength (in length of tape) becomes comparable with the width of the replay head gap, the output from the head will fall, until it reaches zero at the *extinction frequency* where these two are equal. At half this

frequency the output is 3dB down. Equalisation for this high frequency loss is included in the replay amplifier.

In addition to the above needs for equalisation, further losses occur at high frequencies, due to the fall in tape permeability and the tendency for self-demagnetisation to occur within the tape. Again, hysteresis and eddy current losses cause still further high frequency reduction. Equalisation for these is normally included in the recording amplifier.

10.2.5. The Tape Transport Mechanism

As stated in the introduction, the function of this mechanism is to ensure that the tape passes the heads at an absolutely constant linear speed. This is normally achieved by means of a *capstan* driven at a constant speed by means of an electric motor and anti-vibration coupling. The tape is held in contact with the capstan by the *pinch wheel*, a rubber covered pulley held against the tape by spring tension. The driven tape speed may vary from machine to machine, the standard speeds being 30, 15, $7\frac{1}{2}$, $3\frac{3}{4}$, $1\frac{7}{8}$ or even $\frac{15}{16}$ in. per second. Of these, the first two and increasingly the third are used for professional purposes, whilst the last four are at the moment used in domestic equipment. The tape transport mechanism is completed by two further motors arranged to drive the *feed* and *takeup* spools during recording and reproduction and for the rewinding or *spooling* process. On professional equipment and some of the better class domestic equipment, the spooling speed can be varied; on most domestic machines, however, a fixed speed is provided.

10.3. ADVANTAGES AND DISADVANTAGES OF TAPE RECORDING

10.3.1. Advantages

(a) The medium—namely, the tape—may be used over and over again. This is especially useful in a broadcasting organisation where so much recorded material is required only for a short time.

(b) Recording and reproduction may be carried out by remote control from the studio by means of relay-operated circuits.

(c) Editing, which involves cutting and re-joining, or the dubbing of extracts to another tape, can produce a continuous taped programme.

(d) Relatively long playing times are possible on each tape.

(e) Storage is fairly simple, although the length of store life is uncertain.

(f) Many reproductions can be effected without loss in quality or increase in background noise.

10.3.2. Disadvantages

(a) Playback is not possible until the tape is rewound.

(b) A number of recorded passages are less easily superimposed than with discs.

(c) Spurious magnetic fields may introduce noise to tape recordings or partially erase them.

(d) The recorded modulation cannot be *seen*.

10.4. EDITING TAPE RECORDINGS

There are two methods of editing tape recordings. Firstly, when an extra machine is available, the required passages can be dubbed in sequence so as to build up the finished programme. This method has the advantage that it obviates cutting of the original tapes. Secondly, there is the cut-and-join method in which the required passages are cut from the original tapes and a fresh composite tape built up from them. In either case the exact positions of the tape from which the dubbing is extracted or the cuts made are found by moving the tape manually across the replay head at slow speed, listening to the tape to find the required gap. A mark is then made on the tape in the position of the head gap and this is the editing point. Tape editing by the cut-and-join method is not,

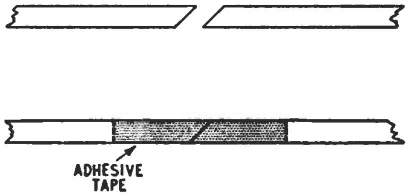


Fig. 10.6. Joining tape

of course, possible when more than one track, each carrying a different programme, is recorded on the tape, without destroying all the tracks except the one being edited.

When a tape is edited by cutting, it is essential that accurate cuts be made so that different pieces of tape can easily be joined together without discontinuities showing when the tape is subsequently reproduced. Some form of editing device is desirable, and

this may conveniently consist of a milled channel in a block of metal in which the tape is laid, the edges of the channel being undercut so that the tape is held firmly. A narrow slot at 45° to the tape channel provides an accurate guide for a razor blade (see Plate 10.2). Joining of tapes may be by means of special adhesive tape applied to the reverse side of the recording tape over a butt joint, or a solvent may be used with a small overlap (Fig. 10.6). Normally the adhesive tape joint is used for normal cut-and-join editing in the studio or editing room, and the solvent used for " service joints " when used tape is checked before subsequent re-issue. Great care should be taken that no adhesive material gets on to the coated side of the tape or it may foul the tape guides, causing faulty reproduction.

10.5. TAPE RECORDING EQUIPMENT

10.5.1. **The Ferrograph Tape Recorder** (Plate 10.3)

This is a semi-portable high grade domestic machine and is principally used by the BBC as a *rehearsal* recorder. The machine is, however, used for transmission by a number of Colonial Broadcasting Stations. Two versions of the machine are available, running at 15/7½ and 7½/3¾ in. per second respectively, and as used by the BBC are provided with input and output circuits for use with zero level programme at 600 ohms impedance. A second input jack is provided for a low level input such as a microphone at high impedance. Bass and treble tone controls are provided, operative on playback only. The machine has no facilities for monitoring the tape while recording.

10.5.2. **The E.M.I. Magnetic Recorder TR/90** (Plate 10.2)

This is a high-quality tape recording and reproducing machine suitable for rack-mounting in a confined space, or for trolley-mounting. Tape speeds of 7½ and 15 in. per second are provided, the latter being the standard speed for BBC programme tapes giving 32 minutes playing time for 2,400 ft of tape. A number of studios have been fitted with special sockets which permit operation of the trolley-mounted TR/90, either for recording or for reproducing to a channel on the desk or via an acoustic effects loudspeaker in the studio.

A spooling control operates when the spool button has been pressed, and gives continuous control over the tape speed, from full speed forward to full speed reverse. Complete rewinding takes

under two minutes. The *run* and *off* controls are relay-operated, so that remote-starting is possible. A *timing indicator* gives a good check on timings, and, being friction-driven, may be set back to zero as required.

Very similar in facilities and performance is the E.M.I. BTR/2, the static tape machine employed in many recording channels.

10.5.3. Leevers-Rich Tape Reproducer (Plate 10.4)

This is a high-quality tape reproducing machine, normally mounted as a transportable console, which can be operated in studios using special sockets as in the trolley-mounted TR/90. Two speeds are again provided, 7½ and 15 in. per second, but unlike the TR/90, this machine is not relay operated from push-buttons. The various functions are obtained by operating the appropriate switches; a large switch on the right-hand side of the machine controls spooling and normal playing, and in the spool position a knob on the left-hand side gives continuous control over the tape speed. Again unlike the TR/90, the change of reproducing speed is accomplished by means of a switch with a knob on the left-hand side of the deck, and a second knob on the panel on the front of the machine must be turned to change over the equaliser in the reproducing amplifier. Both these switches have intermediate "off" positions.

A monitoring amplifier is provided with its own gain control, enabling pre-fade listening either on a small built-in loudspeaker or on headphones. In the normal condition of the machine, the starting time is much slower than the TR/90, but is is possible to arrange that the capstan motor is running continuously, and in this condition the operation is much speeded up. All machines found in studios have the capstan motor running from the time the equipment is switched on.

10.6. REMOTE CONTROL OF TAPE RECORDING AND REPRODUCTION EQUIPMENT

The BTR/2, TR/90 and Leevers-Rich machines are all capable of remote control. Two types of such control are found.

In the first case, the remote facility may be used to start machines in a studio cubicle from the studio desk, the machines having previously been set up, and switched to "remote". This facility is useful when tape inserts have to be cued accurately into fast moving

programmes. Facilities for starting several machines in this way may be provided.

The second type of remote operation involves the setting-up of a central recording room in which recording staff supervise a number of tape machines. Special circuitry at the studio enables the studio manager to stop and start the tape, and to monitor the actual recording. This means he is listening to the programme approximately one-fifth of a second late, which is a delay likely to cause trouble on a fast-moving production, but is tolerable on talks, etc. The P.P.M. reads the normal studio output.

10.6.1. The Broadcasting House Arrangement

In this, all talks and general purpose studios are connected to the central recording room (Plate 10.5), and each is fitted with:

(a) *Record off/run* key and red indicator lamp—used to stop and start machines in the central room, which are set up to *record*.

(b) *Monitor feed*—ring main point 10 is connected to the appropriate machine in the central room, both for record and playback.

Operation of the record *off/run* key also switches on the studio red lights, after a delay of 2 seconds, which has been introduced to prevent the studio starting to speak before the machine has run up to speed.

The arrangements at other centres, such as Bush House, differ in detail from those described above, but the general principles of operation are the same.

10.6.2. Reproduction

If remote controlled reproduction *into studio productions* is required, extra circuitry is necessary to bring up the tape output as a source in the studio. This may be brought to a special fader mounted on a tape control unit, or as with the new Type B control equipment at Bush House, the tape output is automatically switched to the studio on Outside Source line 1.

Reproduction of *complete programmes* into a given Domestic Service is effected in the appropriate Continuity Suite.

10.7. LOCAL RECORDING AT THE STUDIO

An increasing number of studios are being equipped with local recording/reproduction facilities. Being able to record, mix, and edit a programme in step with the producer's arrangements for

rehearsal is useful on many types of programme, particularly since it facilitates the techniques of recording sequence by sequence.

An excellent on-site recording installation consists of three static tape machines with linking console housed in a room adjoining and communicating with the studio control cubicle. A sound-proof window will give good liaison between the rooms, while permitting extra editing or other work to be done as the studio rehearsal proceeds (Plate 10.6).

10.8. PORTABLE RECORDING MACHINES

The use of miniature battery-operated tape recorders for on-the-spot recordings, such as interviews, news reporting and sound effects, has become very popular, and several machines of this type are used.

10.8.1. The E.M.I. Midget Recorder—Type L.2 (Plate 10.7)

This recorder is contained in a wooden rexine covered case, 14 × 7 × 8 in., and two versions are available. The earlier version is valve operated and weighs 14½ lb, including batteries. A separate reproducing head and amplifier allows the recording to be monitored on headphones if required. Playback is also possible, but it is better to playback on static equipment and so conserve the life of the batteries. No erase head is included and clean or pre-erased tapes must be used.

The record/replay switch is on the tape deck, and when this has been put in the required position, and the drive roller latched into position, the lid may be closed, and the machine started by means of the battery *on/off* switch in the end compartment. Three windows in the lid allow the level meter and the amounts of tape cn the two spools to be observed. A geared rewind, operated by hand, is also in the lid.

Nine Venner accumulators or dry cells are used to provide L.T. and motor driving voltage; their life is about 6 hours. H.T. is provided by two 67·5 volt dry batteris whose life is about 15 hours.

The newer version of this machine has a smaller transistorised amplifier and the extra space which is thus available in the case has permitted the inclusion of a monitoring amplifier and loud-speaker.

Input is usually from an S.T. & C. 4032G moving-coil microphone and the 4037 microphone can also be used. The 30 ohm impedance of these microphones matches directly the input circuit of the recorder. The maximum recording time, with a 5 in. diameter reel at 7½ in. per second, is 15 minutes using standard tape.

10.8.2. The Ficord Miniature Recorder—Model 1A (Plate 10.8)

This is an extremely small, transistorised light-weight machine, its dimensions being $9\frac{5}{8} \times 5 \times 2\frac{3}{4}$ in., and its weight $4\frac{1}{2}$ lb. One record/reproduce head is fitted and this means that no monitoring from the tape can be done whilst the machine is recording. Unlike the E.M.I. machine, however, the Ficord has an erase head and so does not need to be used with pre-erased tape. The amplifier is transistorised and a switch converts it from record to play function. A miniature loudspeaker operates on replay for monitoring purposes only. The machine is powered by six lead/acid 2 volt miniature accumulators, each having a sealed plastic case so that the battery is unspillable. A charger is provided having approximately the same dimensions as the recorder, and a fully charged set of batteries will run the machine for two hours at $7\frac{1}{2}$ in. per second. At the alternative speed of $1\frac{7}{8}$ in. per second this battery life is extended to between three and three and a half hours, but this speed is not used for broadcasting purposes.

The microphone used with this machine is a Grampian moving-coil and the playing time for a 300 ft spool of long-play tape is a maximum of nine minutes.

10.8.3. The Nagra Tape Recorder—Model 111B (Plate 10.9)

This is a miniature battery operated portable machine, made to high quality professional standards. The dimensions of the machine are $14\frac{1}{4} \times 9\frac{1}{2} \times 4\frac{3}{8}$ in. approximately, and its weight is 15 lb 11 oz including batteries.

The machine operates at three speeds, 15, $7\frac{1}{2}$ and $3\frac{3}{4}$ in. per second, and a special feature of the design is the excellent speed stability and freedom from wow and flutter. This is largely due to a special electronic stabilising device.

Two inputs are provided, each having a separate fader so that they can be mixed. One is normally for a moving-coil microphone of 50–200 ohm impedance, and the other a high level input at higher impedance. An accessory unit can be obtained enabling a capacitor microphone of the Neumann " KM " type to be used into this high level input, power for the microphone being taken from the recorder. Two outputs are provided, one a monitoring output into headphones or loudspeaker, and the other an output to feed a 600 ohm line at a level of + 6 dB.

The amplifiers are transistorised, and the machine is of the three-head type so that monitoring from the tape is possible during recording.

Level indication is by means of a P.P.M.-type instrument so that good peak indications are given.

The battery consists of twelve 1·5 volt " U2 " type cells, and will give an operating life of 10–12 hours depending on the type of use. The battery voltage is most critical when the machine is operated at 15 in. per second.

The machine will accept 7 in. reels of tape, giving 15 minutes recording time at 15 in. per second or 30 minutes at 7½ in. per second, using standard tape.

Recordings made on this machine can be comparable in every way with those made on static professional equipment.

11

DISC RECORDING AND REPRODUCTION

Disc records, as a means of storing sound, have become a major source of domestic entertainment, and so naturally find their place in a broadcasting organisation such as the BBC. Many programmes are built round commercially recorded discs, and such discs are widely used as incidental music in dramatic and feature productions.

In addition to using commercial gramophone records—the BBC has probably the largest gramophone library in the world—the BBC pioneered the use of the Direct Cut Lacquer disc as a means of recording programmes. Whilst this use of disc recording has now been superseded by tape, a few specialised uses of the medium still remain. For example, in the compilation of News Bulletins, it may be desirable to use only a part of a particular news despatch in a given transmission, whereas in another transmission the whole despatch, or a different part may be needed. The decision as to exactly which extract is needed for a given bulletin may have to be taken after the programme is on the air, and so some rapid means of selection is obviously necessary. Given suitable equipment with discs this can be accomplished in a minimum of time. For similar reasons, sound effects for Features and Drama productions have also been recorded on disc, and a very flexible production routine has been developed over the years, using a number of specially designed turntables running at 78 r.p.m. Modern considerations of quality would dictate that both these types of usage could be bettered by tape recording, but so far no practical tape-reproducing equipment has been developed which will permit the rapid operation and flexibility necessary. In a typical programme, as many as eighty or ninety effects may be necessary, each of only a few seconds duration, and with disc operation, given the specialised equipment, it is possible to find any one of these, and play it into the programme in no more than five

seconds. Montages of a number of effects, each on a separate disc, can be built up, and the composition of a montage can be varied, if necessary, to take up any changes in the tempo of playing a dramatic scene, for example between rehearsal and transmission.

11.1. GENERAL CONSIDERATIONS

In disc recording, the sound vibrations in the air are converted into alternating currents by the microphone. These currents are amplified and sent to the coil of the recording head, where they give rise to an alternating magnetic field. This interacts with the field of the permanent magnet in such a way as to give the armature a turning movement about the pivot rod. Thus, the cutting

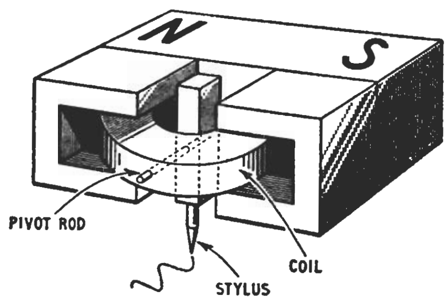


Fig. 11.1. Disc recording head

stylus vibrates to and fro at the frequency of the original sound waves, and will cut a lateral trace on the disc which is rotated at a constant speed (Fig. 11.1).

It is necessary when recording to introduce bass cuts in the programme material in order to prevent the cutting stylus making such a large movement that it tends to cut into the previous groove. It is also desirable to introduce some high frequency lift in order to overcome the worst effects of surface noise when playing the disc.

Some years ago both the bass cut and top lift were different in amount from organisation to organisation and from recording company to recording company, and so correct re-play conditions for any given disc were hard to achieve. In the last few years international agreement has been reached on these *recording characteristics* and standard playback equalisers are now fitted to almost all disc-playing equipment (Fig. 11.2). Discs recorded before the introduction of this standard will not of course be correctly reproduced but the standard has been chosen as a reasonable compromise

and no serious error should result. If, in isolated cases, extra correction is necessary a variable frequency response control unit should be used.

Turntable speeds are measured in revolutions per minute or r.p.m. The speeds normally used are 78, 33⅓, and 45 r.p.m. but recently experimental discs and talking books have been recorded at 16 r.p.m.

78 *r.p.m.* is still used for some commercial gramophone records, and for BBC discs where ease of editing is a prime consideration. Playing times of up to five minutes are possible per side.

33⅓ *r.p.m.* is used for fine groove recording (microgroove recording) as found on commercial long-playing records. It has also been

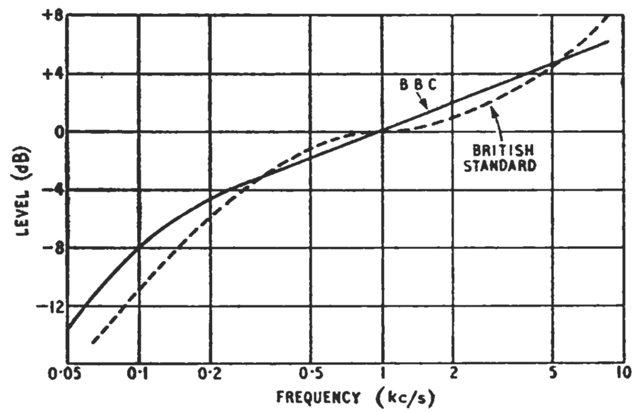


Fig. 11.2. British Standard fine groove recording characteristic

used for Transcription recordings with a coarse groove on a 17¼ in. diameter disc, enabling a 15 minute programme to be recorded without a break.

45 *r.p.m.* is also used for commercial long-playing records.

The number of grooves per inch on a disc is called the *pitch.* 104 grooves per inch are normally cut on BBC coarse-groove discs, although 120 grooves per inch are sometimes used to accommodate long recordings of effects etc. The pitch on commercial 78 r.p.m. records varies a great deal, but about 100–120 grooves per inch is usual. Long-playing gramophone records have shallow grooves at about 220–350 to the inch. BBC fine-groove recordings have a

pitch of 240. In disc recording there is a limit to how closely the grooves can be spaced, and how slowly the turntable can rotate, if good quality is to be maintained.

11.2. DIRECT DISC RECORDING

Direct recording on a lacquer-coated aluminium disc has a number of advantages and disadvantages:—

11.2.1. Advantages

(a) The disc is ready for immediate playback—no rewinding is necessary as with tape.

(b) The rapid editing and fitting together of extracts is conveniently done on discs (this is especially true of 78 r.p.m. discs).

(c) The same turntable desk will, with suitable equalisation, play BBC discs and commercial gramophone records.

(d) Processing, so as to obtain records for permanent retention, or many copies, may be accomplished by the same method as is used in the gramophone industry.

11.2.2. Disadvantages

(a) The recording is not permanent—in fact, very careful handling is necessary if a dozen noise-free playings are to be obtained.

(b) Greater skill and expertise are required to cut a good disc than are necessary to make a good tape recording.

(c) Only a relatively short recording time is possible per disc.

(d) Loss of high frequencies tends to occur towards the centre of the disc.

11.3. PROCESSING

In order to process a direct recorded disc, a master or matrix must be made, from which, by further processes, any number of copies of the original recording can be produced.

The principal material for the making of pressings was previously shellac, but more recently vinyl has been introduced. This gives lighter and less brittle pressings, with less surface noise. Handling and storage of these new pressings calls for a great deal of care, however, as they attract dust, and are marred by even tiny scratches.

In full-processing, the *master* is obtained by immersing the prepared original in an electro-plating bath and "growing" a

negative of pure copper on to it. From this master, the *mother* is obtained by another electro-plating process, and will have grooves identical to those of the original. The *stamper* is prepared from the mother, and makes a " positive " impression on the final record or pressing, which is malleable at high temperatures, and becomes hard when cooled (Fig. 11.3).

Half-processing is undertaken when only up to 50 pressings are required. The master also becomes the stamper, after being hardened by nickel and chromium plating. There is naturally a



Above: Fig. 11.3. Steps in full-processing.



Left: Fig. 11.4. Steps in half-processing

saving of time and expense, and many BBC discs are duplicated in this way (Fig. 11.4).

11.4. TURNTABLE DESK TD/7

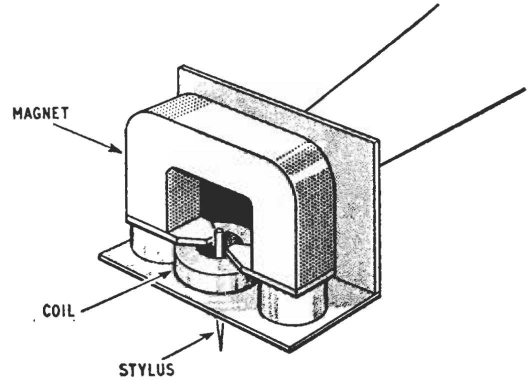
This desk was developed some years ago for the playing of BBC 78 r.p.m. disc recordings, as well as commercial gramophone records. It is being retained for as long as is necessary in accordance with production requirements. It is a twin turntable desk, and with it the playing of short extracts from any part of a 78 r.p.m. disc is possible with considerable accuracy. It will be described in detail.

11.4.1. **Reproducing Head Type E.M.I. 12** (Fig. 11.5)

This is a medium-weight reproducing head fitted with a sapphire-tipped stylus. A counter-balance gives a tracking weight of 1½ ounces.

The stylus fits into a tubular armature and is held through slots by an elastic band and by magnetic attraction. It is important that the needle should not be rotated, as a " flat " may have been worn which will damage the disc. For this reason, and to prevent

Fig. 11.5. E.M.I. Type 12 reproducing head



unauthorised removal, the needle is sometimes cemented into the armature tube. By modern standards, this pick-up is no longer considered " light-weight ", and any future head is likely to be much smaller.

11.4.2. Equalisation

A two-position key has been fitted to meet the different requirements of BBC and commercial recordings. In the *direct recordings* position, the pick-up is connected to the output circuit via an equaliser circuit which corrects for the BBC recording characteristic which rises at about 4 dB per octave (Fig. 11.6). In the *gramophone*

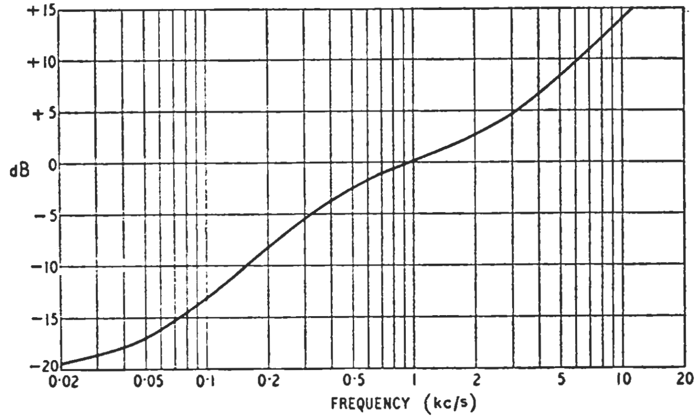


Fig. 11.6. 78 r.p.m. recording characteristics

records and pressings position, further attenuation is introduced above 6,000 c/s to reduce surface noise, and the overall level is brought down 4 dB to correspond to the level recorded on BBC discs (see Fig. 11.7). A step-by-step attenuator (0–6 dB in 2 dB steps) is

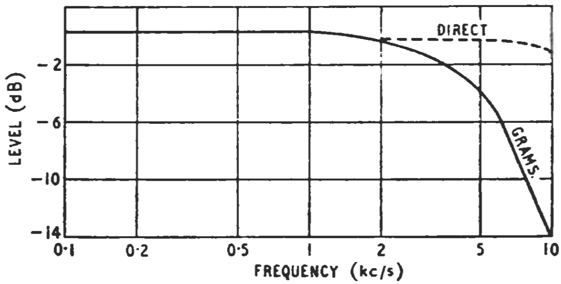


Fig. 11.7. Effect of 2-position key on TD/7

provided inside the desk, and is adjusted by the engineering staff to compensate for the varying sensitivities of different heads.

11.4.3. Pre-fade Listening

As may be seen from the block diagram (Fig. 11.8), a separate output is taken to each pre-fade amplifier. A separate two-position key for each turntable switches the headphone circuit either to the pre-fade amplifier or to the programme selector switch.

11.4.4. Stroboscope

When the turntable is rotating at its correct speed, the beams of light from the neon lamp appear to be stationary. This is because the number of holes round the turntable has been calculated so that each hole is exactly replaced by the next, in step with the alternating brightness of the a.c.-driven neon lamp. Now, with 50 c/s mains, the neon glows bright 100 times per second, or 6,000 times per minute. If, therefore, 6,000 holes pass the neon per minute, the pattern will appear stationary, and for 78 revolutions per minute there must be 6,000/78—i.e. 77 (approximately) holes equally spaced round the edge of the turntable.

For the same reason, 77 green bars are printed round the edge of the labels on BBC discs, giving a useful stroboscope indication when looked at in light from 50 c/s mains.



Plate 11.1. Turntable desk TD/7 showing DETU/1

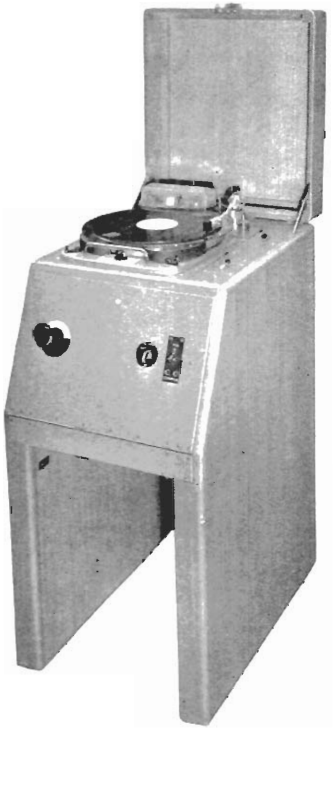


Plate 11.2. BBC fine-groove reproducing desk DRD/5

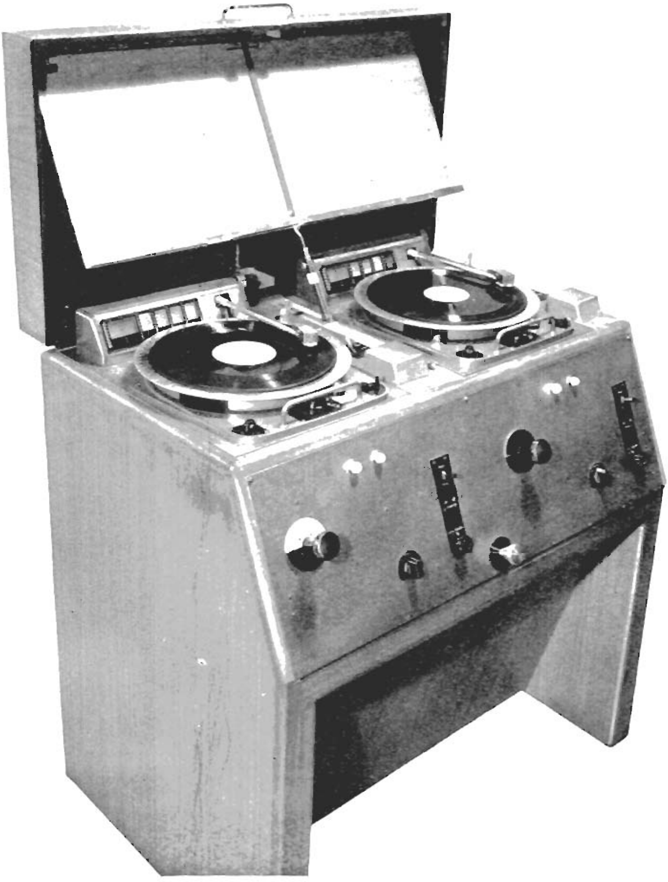


Plate 11.3. BBC disc reproducing desk RP2/1

However, if the mains frequency is not exactly 50 c/s at the time of reproduction, perhaps due to a power-cut, setting by the stroboscope will give wrong speed adjustment. It is necessary, then, to play a record of 1,000 c/s tone on each turntable in turn, and adjust the speed until the pitch lines up with tone sent by the control room (for perfect speed adjustment beats between the two tones should be eliminated). Recording staff are instructed to record a short band of tone at the beginning of at least the first two discs of a programme, and this tone should be synchronised with the control room tone a short time before transmission.

11.4.5. Groove-locating Unit GLU/9B

Parallel tracking is used, which simplifies the design of an accurate groove-locating mechanism, but has so far prevented the introduction of an up-to-date light-weight pick-up. The pick-up head is attached to a light-weight tracking-arm which is held by a carriage. This runs on a polished carriage-rail, and its six ball-races ensure minimum friction and side-play. The rail itself is supported in ball-races, within anti-vibration mountings.

Operating the lifting lever causes a bar to press down the counter-balance, so raising the pick-up. A drum at the end of the carriage-rail gives fine adjustment of the pick-up position. A scale with

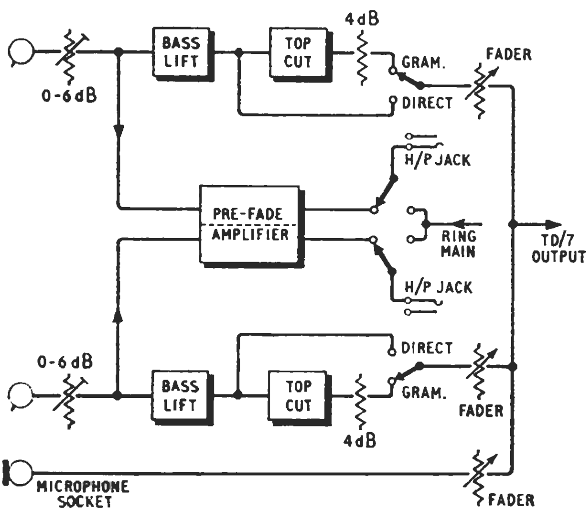


Fig. 11.8. TD/7—circuit diagram

divisions, each representing ten grooves at a pitch of 104 grooves per inch, is fixed to the cover of the GLU, and this helps to locate narrow bands on the disc.

The simplest procedure for groove-locating is described in this quotation from the BBC Recording Training Manual, which should be consulted for a more detailed description of the apparatus:

> " The disc is played until the beginning of the required excerpt is heard on headphones connected to the pre-fade amplifier; the pick-up is then raised by the lever, the turntable continuing to rotate; the fine control is then turned so as to set the pick-up back by a small amount. The precise setting of the fine control is a matter of practice rather than skill and the accuracy obtained is sufficiently high for most requirements. A little thought will show that the application of this system to 33⅓ r.p.m. discs would be less satisfactory, because of the increased programme time occupied by a single groove ".

As is well-known, the best accuracy obtainable, using the simple procedure described above, is plus or minus half-a-groove. To improve on this, it is necessary to halt the pick-up in a particular part of the required groove. Doing this by hand necessitates braking the turntable suddenly and spininng it on cue. Unless this is done very gently, the sapphire digs in and puts a click on the disc. If it is done roughly, damage is possible to the turntable motor (not to mention the disc), and in particular a great strain is put on the anti-vibration coupling. The motor driving-shaft is coupled to the turntable spindle via two plastic discs which are riveted together. This virtually isolates the pick-up from motor vibrations, but, of course, makes gentle handling necessary.

11.4.6. Quick-start Devices

The necessity for accurate cueing of disc inserts, and the ever-increasing speed of operation has necessitated the introduction of a quick-start device for 78 r.p.m. turntables.

In the original device, the operation was by means of two push-buttons, one red, one green.

Pressing the red button causes a pair of rubber-covered blades to press against the underside of the disc, thus lifting it just clear of the turntable, and bringing it to rest almost instantaneously. The pick-up is still resting on the disc, and the setting-up procedure is simply one of arranging that the disc is stopped about half-a-revolution before the desired cue.

Pressing the green button causes the lifting blades to fall away, and the disc to drop back on to the already revolving turntable. As soon as the relatively light disc touches the rubber friction mat of the turntable, it starts to rotate and quickly reaches the correct speed. With practice it is possible to fade up without any trace of " wow ", and start the disc at any required cue.

The mechanical action is simple, and may be appreciated by reference to the rough diagram in Fig. 11.9. Pressing the red button causes the pivoted lever to raise the lifting blades until the lever is

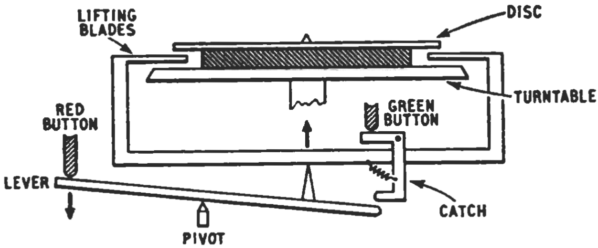


Fig. 11.9. Prototype quick-start device

engaged by the catch. This holds the disc stationary and clear of the turntable. It is possible to " edge " the disc round by pushing the red button just short of the " engaged " position.

Pressing the green button disengages the catch, so that the lifting mechanism and disc return to their lower position, and the disc is accelerated quickly to normal speed.

An improved quick-start device has been produced by BBC Designs Department, and given the code name DETU/1—" Drop-start Editing Turntable Unit ".

The red and green buttons are replaced by a single lever, and the two lifting pads are replaced by a three-point lift. The original 12-in. turntable is replaced by one of 9 in. diameter, which permits a clearance for the lifting mechanism for 10 in. discs. A paper strip round the edge of the turntable carries the stroboscope markings, and is illuminated by a pair of neon lamps. A new speed-control knob has been fitted—see bottom left-hand corner of photograph (Plate 11.1).

11.5. PRESTO REPRODUCING DESK

This two-speed desk is used by the BBC mainly for the reproduction of discs recorded at 33⅓ r.p.m. The development of tape

recording for many programmes has considerably reduced the use of " slow-speed " discs. A rubber tyre is fitted round the rim of the turntable and the driving roller, which presses against this, has two diameters. The smaller of these drives the turntable at 33⅓ r.p.m.,

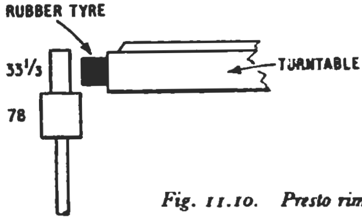


Fig. 11.10. Presto rim-drive mechanism

and the larger at 78 r.p.m. The appropriate speed is obtained by operating a speed-selector lever (Fig. 11.10).

After use, the locking-nut must be loosened, and the lever moved to the left to remove the drive pressure. Failure to take this precaution may result in flats being impressed on the turntable tyre, with consequent " wow " and rumble.

This desk is normally used to reproduce 17¼ in. slow-speed transcription recordings, having a coarse groove, although some desks have been modified with an additional pick-up for fine-groove long-playing discs.

11.6. FINE-GROOVE REPRODUCING DESK DRD/5 (Plate 11.2)

This desk is a specially designed turntable and pick-up for reproducing fine-groove records at speeds of 33⅓ and 45 r.p.m.

A lightweight crystal pick-up is provided and there is an optical system associated with the pivot of the pick-up arm which gives an approximate indication of the position of the pick-up head on a ground glass scale at the back of the instrument. The accuracy is such that resetting to a given division on the scale ensures that the pick-up is within five grooves of the passage on the disc.

A sensitive quick-start device is provided; in this case the turntable rises to meet the disc rather than the disc dropping, as in the case of the TD/7. The speed of starting is such that the disc is running accurately to speed within one-quarter of a revolution. An improved stroboscope is provided consisting of a large plate revolving on the turntable having the stroboscope markings provided by means of holes drilled round the edge: these are viewed by means of a neon lamp which has a special pulse sharpening circuit to enable

a clear indication to be obtained. Indicating lamps are provided to show whether 45 r.p.m. or 33⅓ r.p.m. has been selected and a red indicator panel is illuminated when the fader is turned up.

Pre-fade listening on headphones is available, and also a top cut filter providing a tail-off above 10 kc/s, which is useful in the removal of distortion which is sometimes present on worn discs.

11.7. DISC REPRODUCING DESK RP2/1 (Plate 11.3)

The RP2/1 is a high quality desk for the playing of all types of disc with the exception of 16 in. slow-speed coarse-groove types. Hence it will play all 78 r.p.m., 45 r.p.m., and 33⅓ r.p.m. coarse or fine-groove records.

In contrast to the DRD/5, the RP2/1 has two turntable and pick-up units mounted in the same console cabinet. Each unit has its own equaliser pre-amplifier, powered from a common power supply. The quick-start system of the DRD/5 has been retained, but is now electric motor operated, and, in consequence, has a slight time delay. The optical groove indication is also retained, and there is a simplified raise/lower mechanism for the pick-up arm.

Pre-fade facilities are provided, and a pre-fade gain control has been added giving a boost of some 10 dB if required.

The pick-up head is a magnetic variable reluctance type of the turn-over variety, allowing for change of stylus for coarse and fine-groove discs. The equalisation for the two types of disc is changed automatically with the head by means of a miniature micro switch in the head itself, which operates a relay on the amplifier units. A cross-linkage with the turntable speed switch causes the speed indicator lamps to flash if the stylus selected is unsuitable to the disc speed, but this does not prevent the disc being played in this condition. Care must, therefore, be taken to ensure that fine-groove discs are not played with the stylus for coarse grooves, or damage to the disc will result.

12

THE BROADCASTING CHAIN AND DISTORTION

12.1. INTRODUCTION

The description which follows is concerned with the Broadcasting Chain as developed by the BBC, and is especially suited to the type of sound broadcasting found in the United Kingdom where one basic organisation, together with regional stations, supplies programmes covering the whole country. Other organisations may find different systems more suitable to their requirements, but many of the individual stages in the process described below will be of interest.

The sequence of events through which a BBC programme passes on its long journey from the studio to the listener's home is as follows (Fig. 12.1):—

(a) The sounds in the *Studio* are picked up by the microphone, and converted into an equivalent low level electric current.

(b) This is mixed with other sources where necessary in the *Control Cubicle*, amplified and passed to continuity.

(c) In the *Continuity Suite* the output of the studio is mixed with other sources or interspersed with continuity material, and passed to the control room.

(d) In the *Control Room* the programme is suitably amplified and passed to the appropriate land line.

(e) The *Post Office Lines* carry the programme to one of more transmitting stations.

(f) At the *Transmitting Station*, the programme is amplified, combined with radio frequency carrier currents (the process known



Plate 12.1. Light Programme continuity suite

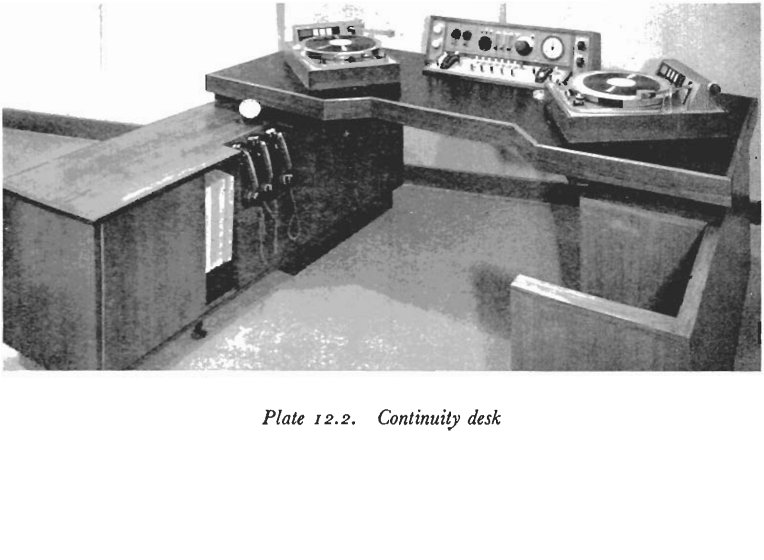
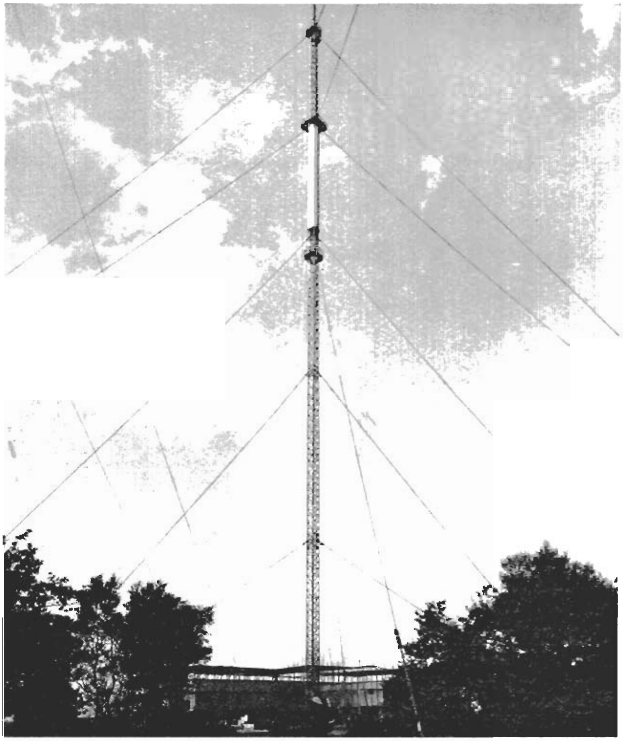


Plate 12.2. Continuity desk



Above: Plate 12.3. The control room. Below: Plate 12.4. Pontop Pike transmitting station during construction, showing the tubular section of the main TV mast for radiating VHF sound transmissions



as modulation) and fed to the transmitting aerial which radiates the combined signal.

(g) The signal is picked up by the receiving aerial and passed into the *receiver* where the programme current is removed from the carrier (the process known as demodulation or detection), amplified, and fed to the loudspeaker which vibrates at the same frequencies as the original sound in the studio.

The studio manager's responsibility for the programme may be taken to end at the point where it leaves the studio. It is felt,

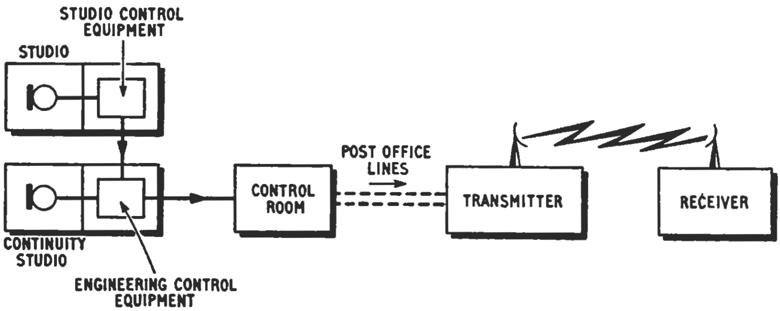


Fig. 12.1. The programme chain

however, that a brief description of the functions and difficulties of subsequent links in the chain will provide useful extra knowledge. Following this analysis of the broadcasting chain there will be a discussion of its imperfections as a means of producing faithful reproduction of programmes in the listeners' homes.

12.2. THE CONTINUITY SUITE

The continuity engineer is responsible for technical monitoring of the programme as a whole, and initiating engineering action in case of fault. It is also his duty to select subsequent source circuits and carry out routine tests on each, prior to transmission. Finally, he exercises control on the volume of fill-ups and other contributions from the continuity studio. Studio and outside broadcast items are, of course, controlled at source, and no subsequent control is normally exercised by the continuity engineer (Plate 12.1).

Continuity working was introduced in the BBC in 1942, and the layout of the control equipment has been modified only slightly since that time. The various sources—studios, reproducing rooms, etc.—appear on a series of source jacks. These are selected one by one as the day progresses, and plugged to one of the input jacks

associated with a four-channel mixer. The output of the mixer is connected to an amplifier, which feeds the main programme to the control room.

The desk also incorporates pre-fade and programme meter facilities which are used during the pre-transmission tests on each source (Fig. 12.2).

The equipment in the continuity studio adjoining the continuity cubicle includes a microphone and at least two gramophone turntables complete with fader controls. There is also a main fader

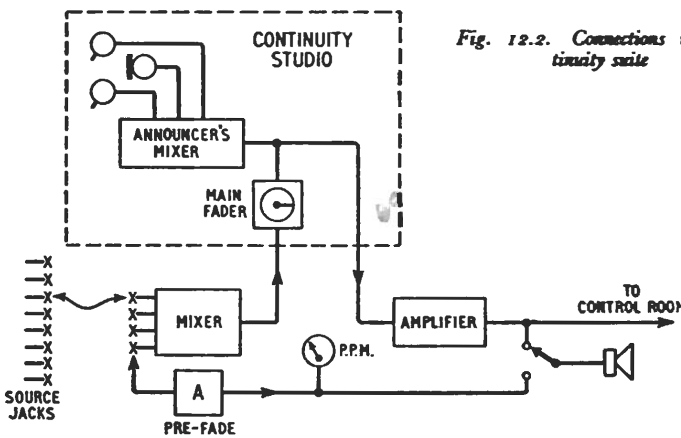


Fig. 12.2. Connections in continuity suite

which allows the announcer to fade down the main programme to superimpose announcements, or to fade it out completely because of technical faults or unsuitable programme material. The announcer normally listens to the main programme continuously on a loudspeaker which is automatically silenced when the microphone is live. He can switch the loudspeaker or headphones to the faulty contribution, while filling up, to check when conditions are favourable to rejoin it (Plate 12.2).

12.3. CONTROL ROOM (Plate 12.3.)

The control room engineers receive programme contributions from a variety of sources, and direct them to the appropriate destinations. This involves a host of incidental operations, including the setting up and testing of routes according to the programme schedule, monitoring miscellaneous programmes, and keeping a

check on the various circuits in order to trace and correct faults with the least possible delay.

The control room is also a kind of switchboard for communications and telephone connections between all points from programme sources to destinations. The staff of the control room also carry out the day-to-day testing of all studio equipment, as well as the equipment in the control room itself, and deal with any faults and technical problems which arise in rehearsals, etc.

12.4. THE POST OFFICE LINES

The network of programme lines interconnecting the various studio centres and transmitting stations of the BBC form what is called the Simultaneous Broadcast (S.B.) System (Fig. 12.3). These circuits are more or less permanent, and are supplemented on a short-term rental by special circuits to meet Outside Broadcast and other commitments. The routing of contributions into the Light Programme, for example, is continually changing throughout the day, whereas the outward connections to the Light Programme transmitters remain fixed. The focal point in this network is the Light Programme continuity suite in Broadcasting House (see map).

A contribution from outside London, say from Belfast, is routed via intermediate BBC stations on S.B. " links ", as they are called. Similarly, the output from Continuity in London travels back to the Northern Ireland Light Programme transmitter at Lisnagarvey. There are nine links in this complete chain.

The intermediate stations do not simply pass the programme on as it is received. Loss in volume takes place along the line, and to preserve a satisfactory ratio of programme to noise (better than 40 dB) amplification is necessary. A further complication arises from the fact that the line loss is not the same at all frequencies. In fact, each section of line has a characteristic response, depending on its length, temperature, etc., which may be different from any other. The only characteristic which most lines have in common is that loss tends to be greater at high frequencies than low. In addition to the *amplifier* inserted at intermediate and receiving stations, it is therefore necessary to include an *equaliser*.

An equaliser is an electrical circuit made up from resistors, capacitors and inductors, designed to give as near as possible the mirror image of the response of the line. The effect will be to restore the original quality before passing the programme into the next section of line. When the distance between BBC stations is more

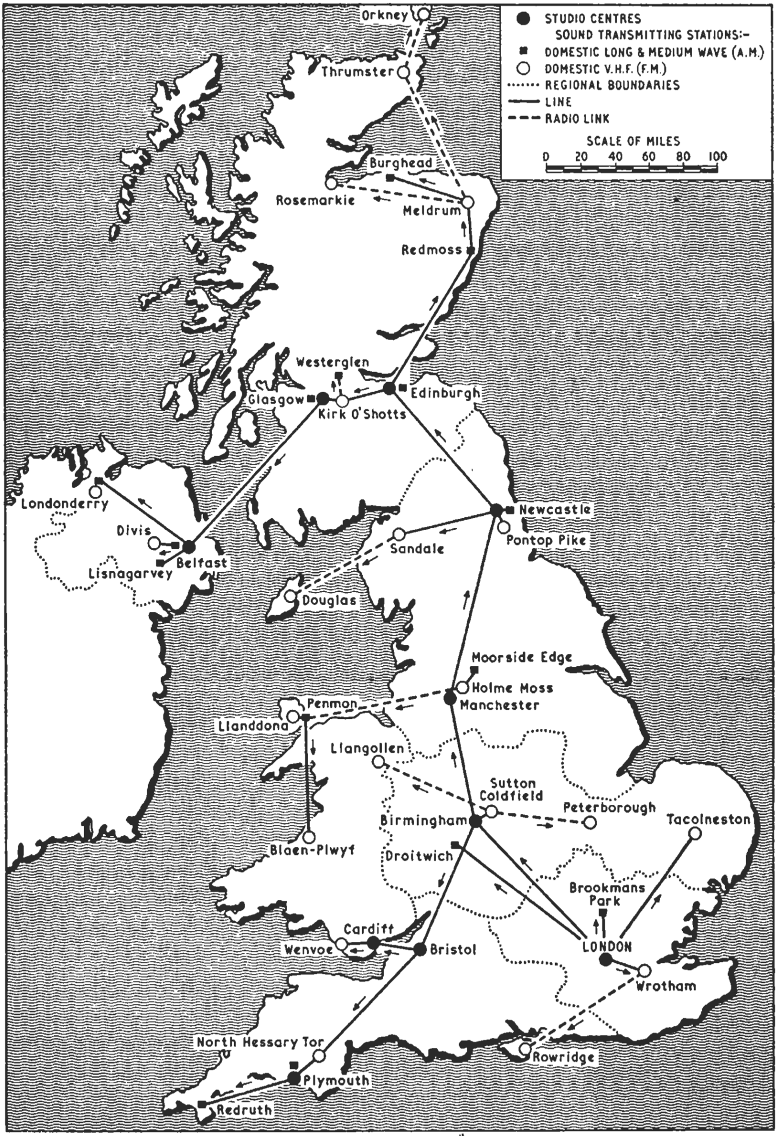


Fig. 12.3. Distribution networks for BBC Light Programme

than a few miles, it becomes the responsibility of the G.P.O. to carry out such amplification and equalisation at intervals as may be necessary. The stations set up for this purpose are known as *repeaters*.

12.5. THE TRANSMITTING STATION (Plate 12.4)

On arrival at the transmitting station, the programme is suitably amplified, equalised, and fed into the transmitter itself. As might be expected, the transmitter valves and other components are physically much larger than their counterparts in studio equipment, since the power developed by the transmitter may run into hundreds of kilowatts—compared with zero level, the normal sending level in studio centres, which is 0·001 watts. It follows that a number of special problems are met in transmitters, such as insulation against the very high voltages used, provision of water cooling systems to dissipate the enormous heat generated by the large valves, etc.—but there is not space to describe these in detail. Instead, two aspects of transmission only will be discussed.

1. Propagation characteristics on the different wavebands.
2. The relative merits of amplitude and frequency modulation.

12.5.1. Propagation of Radio Waves

We have seen in Chapter 4 that a wire carrying current is surrounded by a magnetic field. Further, passing an alternating current through the primary winding of a transformer results in an e.m.f. being set up in the secondary. If this kind of " action at a distance " could be extended over miles, and even thousands of miles, world-wide transmission of signals would be possible.

Unfortunately, the problem is not as simple as the above remarks might suggest. Early experimenters in wireless transmission found that the radiation from an aerial carrying current at studio frequencies (a few thousand cycles per second at most) was very weak indeed. It is only when current alternates at what are now called *radio frequencies*—15 kc/s up to about 3,000,000 Mc/s or more—that energy is thrown off from the aerial in such a way as to travel over long distances.

Radio waves possess electric as well as magnetic properties, and are therefore called *electromagnetic waves*. They travel in free space at a velocity of 300,000,000 metres per second (about 186,000 miles per second), which is the speed of light. In fact, they are similar to light waves in every way, except that they are of much longer

wavelength. To calculate the wavelength of a radio wave whose frequency is known, we have recourse to the formula which we used earlier for sound waves:—

$$c = f\lambda$$

$$\therefore \quad \lambda = \frac{c}{f} \text{ or wavelength} = \frac{300{,}000{,}000}{\text{frequency}}$$

where frequency is in c/s and wavelength in metres.

The wavebands in use for broadcasting have been divided for convenience as shown in Table 12.1.

The propagation characteristics are found to vary considerably for the different wave bands (Fig. 12.4).

The *ground wave* is the name given to the direct ray which leaves the aerial and travels along the surface of the earth. The distance over which the ground wave will carry effectively is found to fall

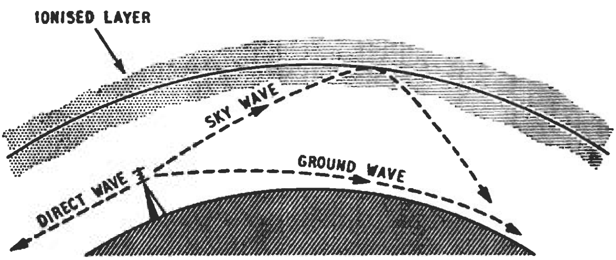


Fig. 12.4. Propagation diagram showing (a) ground wave; (b) sky wave; (c) direct wave

from some thousands of miles on the Long waveband to about 100 miles on the Medium waveband, dependent upon the power radiated. Further reduction in wavelength continues to bring down the range of the ground wave, until at V.H.F. the ground wave is virtually non-existent and reception is only possible by the direct wave at points which have a " line of sight " view of the transmitter. This is usually a maximum of about 50 miles, for normal transmitting and receiving aerial heights, depending on the power radiated, and also on whether there are obstructions such as hills or large buildings in the path of the transmission.

The *sky wave* is the name given to the indirect ray which leaves the earth's surface at an angle, and only returns after reflection from

Table 12.1.

WAVEBANDS USED FOR BROADCASTING

Name	*Wavelengths*	*Frequencies*
Long waves	above 1,000 metres	150–285 kc/s
Medium waves	200–500 metres	525–1,605 kc/s
Short waves	10–100 metres	8 bands in range 6–26 Mc/s
V.H.F.—band I	7·3–4·4 metres	41–68 Mc/s
V.H.F.—band II	3·4–3 metres	87·5–100 Mc/s
V.H.F.—band III	1·7–1·4 metres	174–216 Mc/s
U.H.F.—band IV	0·6–0·5 metres approx.	470–582 Mc/s
U.H.F.—band V	0·5–0·3 metres approx.	606–960 Mc/s
Microwaves		11·7–12·7 Gc/s

the layers of ionised gas (the ionosphere) which exist at heights of about 60–100 miles above the earth.

Little use is made of the sky wave for transmissions on the Long waveband. On medium waves, use is sometimes made of the sky wave but it is something of an embarrassment since it may be returned to earth at points where the ground wave is still fairly strong. The different distances over which the waves have travelled may cause them to arrive out of phase, so there is a region in which *fading* takes place due to interference between the two signals. Reflection of medium and long waves by the ionosphere only occurs after dark, so this interference is more likely to occur at night.

It is on the Short waveband that the sky wave is used extensively, and the radiation is purposely angled to the horizontal to achieve a *skip* distance appropriate to the required destination. It is not proposed to go into detail on the many problems of short-wave

Table 12.2.

TYPES OF WAVEBANDS AND THEIR USES

Wave band	*Type of Wave*	*Principal Uses*
Long waves	mainly ground wave	long range broadcasting
Medium waves	ground wave and sky wave	short range broadcasting long range broadcasting (at night only)
Short waves	sky wave	world-wide broadcasting
V.H.F. band I	direct wave	television—channels 1–5
V.H.F. band II	direct wave	f.m. sound broadcasting
V.H.F. band III	direct wave	television—channels 6–13
U.H.F. band IV	direct wave	yet to be developed for television
U.H.F. band V	direct wave	
Microwave band	direct wave	yet to be developed

propagation, except to say that for a given sending angle, there is a limiting frequency above which waves penetrate the layers of the ionosphere and are not returned to earth. Again, the effective height of the ionised layers is continually varying with seasonal and hour-to-hour changes in the sun's position. The choice of a suitable wavelength for transmission to New Zealand, for example, will be extremely complicated, since of the several hops (reflections) between the earth and the ionosphere which are necessary, if the first takes place at midday in midsummer, the last will take place at nearly midnight in midwinter.

At very high frequencies, the *ground* wave is very quickly absorbed, and no predictable reflection of the *sky* wave takes place from the ionosphere. Reliable transmission is only possible, therefore, over optical distances with no obstructions.

The above information is summarised in Table 12.2.

12.5.2. Modulation

So far we have been concerned with the transmission of the radio frequency wave. But the actual information that is to be broadcast is contained in the programme currents at audio frequencies. It is now necessary to examine the methods used to impress this low frequency information onto the radio frequency " carrier " signal—the process earlier called *modulation*. There are two systems of

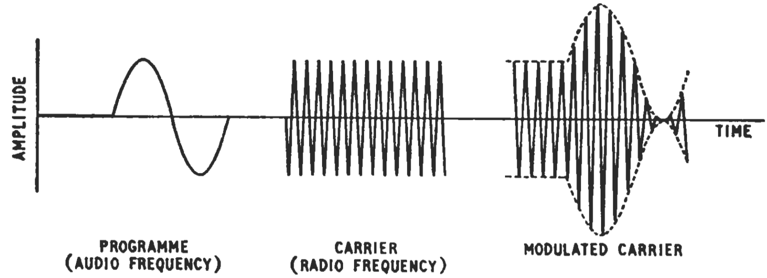


Fig. 12.5. Amplitude modulation

modulation in common use, amplitude modulation and frequency modulation.

(a) In *amplitude modulation* (a.m.), the *amplitude* of the carrier current is made to vary in accordance with the amplitude of the programme current, the number of variations per second being

equal to the programme frequency. The process is shown diagrammatically in Fig. 12.5, where the audio signal is seen to appear as the *envelope* of the carrier wave.

The degree of modulation is the ratio of the audio and radio amplitudes expressed as a percentage. For example, if the a.f. amplitude is half the r.f. amplitude, we have 50% modulation.

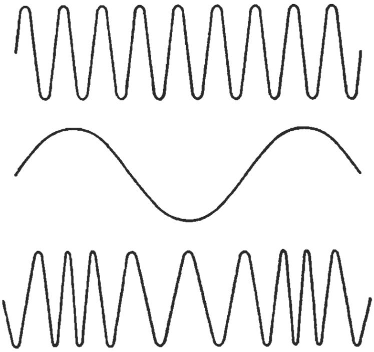


Fig. 12.6. Frequency modulation

The maximum possible degree of modulation is 100% which results in the combined amplitude rising to twice the r.f. amplitude and falling to zero in alternate half-cycles. Amplitude modulation is used in television broadcasting, and until recently was used for all sound broadcasting in this country.

(b) In *frequency modulation* (f.m.), the *frequency* of the carrier current is made to vary in accordance with the programme amplitude, the process taking place as in a.m. at the programme frequency. In BBC f.m. sound transmissions a frequency swing of $\pm$ 75 kc/s is taken to correspond to maximum modulation (Fig. 12.6). (Compare 100% modulation in a.m.)

It follows that f.m. transmissions are invariably placed in the V.H.F. bands where the swing to either side of the carrier frequency can take place without crowding by adjacent stations. The V.H.F., f.m. sound transmissions of the Home, Light and Third programmes are now available to practically the whole of the population of the United Kingdom.

In addition to the freedom from interference by adjacent stations achieved in f.m. transmissions, there is the important advantage that electrical interference from refrigerators, hair driers, etc. (a great nuisance on a.m. reception in large towns) is much reduced,

due to the fact that the f.m. receiver is insensitive to the mainly amplitude modulation of this type of interference.

V.H.F., f.m. transmission and reception also enables a higher quality of sound reproduction to be achieved.

12.6. THE RECEIVER

A detailed description of receiver circuits is beyond the scope of this handbook.

12.7. FIDELITY IN REPRODUCED SOUNDS

It is impossible to achieve completely faithful transmission or recording of programmes. If ideal reproduction of the sounds was possible, the listeners would receive the sounds with exactly the same degree of realism as if they were present in the studio. Fortunately, it is possible to obtain adequate reproduction within the limits of the system for most purposes.

Consider the following list of ways in which reproduced programmes can depart from true fidelity:

(a) The *loudspeaker size* is fixed, and yet it takes the place of all programme sources from orchestras to news readers.

(b) The *monophonic system* of broadcasting (a single channel feeding a single loudspeaker) deprives us of the ability to locate the directions of sounds.

(c) The *loudspeaker volume* will usually be different from that of the original.

(d) The *acoustics of the listening room* will be added to that of the original.

(e) The *dynamic range* is compressed to less than 30 dB.

(f) The *microphone balance* is subject to the taste and experience of the individual studio manager or engineer.

(g) *Noise* is generated in the programme chain.

(h) *Non-linear distortion* tends to be generated in the programme chain.

In this list, which may not be exhaustive, (a), (c) and (d) are seen to be beyond the control of the originators of the broadcasts or recordings. Methods of reducing the harmful effects of (e) will be discussed in Chapter 14, and (f) is largely a matter of careful selection and training of balance personnel.

Something can be done by engineering and S.M. staff to minimise the effects of (g) noise, and (h) distortion, and the varieties of these which can be met will now be briefly summarised.

12.7.1. Types of Noise

(a) *Acoustic noise* in the studio originates from a number of causes and is reduced by careful sound-proofing, carpeting, etc.; it is always high in television studios, which is one of the reasons for using cardioid microphones.

(b) *Amplifier noise* takes the form of a steady hiss. It is caused by the impinging of the electron stream on the anodes of valves, and by the random movements of electrons in the amplifier components.

(c) *Low frequency hum* is associated with the a.c. mains supply. It is at the mains frequency—50 c/s—and possibly harmonics, and results from faulty rectification of the supply, faulty screening, or placing microphones, etc., too near electrical apparatus.

(d) *Line noise* is of many different kinds. Examples are cross-talk due to pick-up from other Post Office circuits, and random thermal movements of electrons. The latter effect is present even in short wires.

(e) *Receiver noise* in the listener's set includes interference from other stations, static from electrical apparatus, and usually some mains hum. Listeners to the V.H.F. f.m. transmissions experience much less noise from machinery etc., and are very seldom troubled by interference from other stations.

It is important for studio managers and engineers to remember that they monitor programmes at a point of low noise level, compared with listeners, and this should be taken into account when setting the dynamic range and the relative level of sound effects, etc.

12.7.2. Types of Distortion

In general terms, distortion is said to be present when any change in waveform takes place between two points in a transmission system. Of the various forms of distortion, attenuation distortion is discussed here at some length, while only definitions are given for other types.

(a) *Phase distortion* is present when the transmission time through the system is different for different frequencies. It is not as a rule

serious at audio frequencies, except on very long line circuits where it may distort transient sounds in speech and percussive music.

(b) *Non-linear distortion* is present when the properties of the system vary for different levels of programme input. It will be introduced by circuit components which have response characteristics which change with input amplitude—e.g. an amplifier when overloaded or a transmitter when over-modulated. Non-linear distortion may be subdivided into the following varieties, all of which are not necessarily produced together:—

Amplitude distortion—variation of gain with input volume such as may occur in a limiter at the instant of overload.

Harmonic distortion—production in the output of harmonics of the input frequencies.

Intermodulation distortion—production in the output of sum and difference tones of the input frequencies.

The effect of the last two forms of distortion is to produce a harsh quality, and to distort the timbre of musical instruments.

(c) *Attenuation distortion* (sometimes called *frequency distortion*) is present when the properties of the system vary with frequency. It

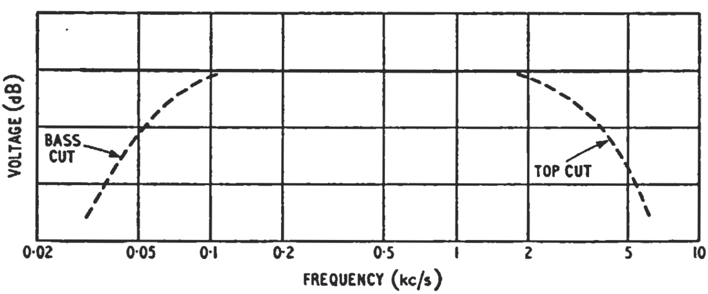


Fig. 12.7. Two forms of attenuation distortion—top cut and bass cut

may be introduced at any point in the programme chain—studio, microphone, amplifiers, lines, transmitter, receiver or loudspeaker —where the response to frequencies from 50 c/s to at least 10,000 c/s is not uniform.

Loss of high frequencies (top cut) reduces the intelligibility of speech and the " brightness " and " attack " of music. Loss of

low frequencies (bass cut) produces a thin quality and lack of weight. When only a narrow band of frequencies is transmitted—both top and bass cuts present—a telephone-like quality results. Boosting of certain frequencies also causes a departure from true quality (Fig. 12.7).

Provided attenuation distortion at a given point in the chain is not too acute, it is possible to pass the programme through an equalising circuit (see Section 12.4) and restore the original quality.

13

THE PLACING OF MICROPHONES

This chapter summarises standard practices in the positioning of microphones. It is intended to help newcomers, and form a basis of experiment for more experienced studio managers. These notes are based on the experience of many people in balancing programmes, but cannot, of course, take the place of actual experience. Where there are drawings or photographs of layouts it is essential to regard these only as examples chosen from the many possible solutions to the given problem—e.g. the final position chosen for a piano microphone will depend on the studio, the player, the particular piece of music, and even the particular piano, and cannot be rigidly laid down.

Two-sided ribbon microphones are considered throughout, except where stated. Other types of microphone, cardioid and omnidirectional may be used under suitable circumstances.

Before considering some of the particular cases that will arise in the positioning of microphones it is necessary first to consider in general the effect to be expected when more than one microphone is used at once.

13.1. INTERFERENCE EFFECTS IN MIXING

The use of more than one microphone can give rise to undesirable effects, since the microphones will tend to interfere with each other.

13.1.1. Special Case of Two Microphones Equidistant from Source

If a sound source is placed equidistant from two microphones, and identical terminal connections are made to two channels of a mixer, the resultant currents in the mixer will be in phase—i.e. will be " in the same sense ". It follows that they will *add* at the input to the amplifier (see Fig. 13.1(a)).

If one of the microphones is now reversed (see Fig. 13.1(b)), or terminals 1 and 2 interchanged, the diaphragm movements will

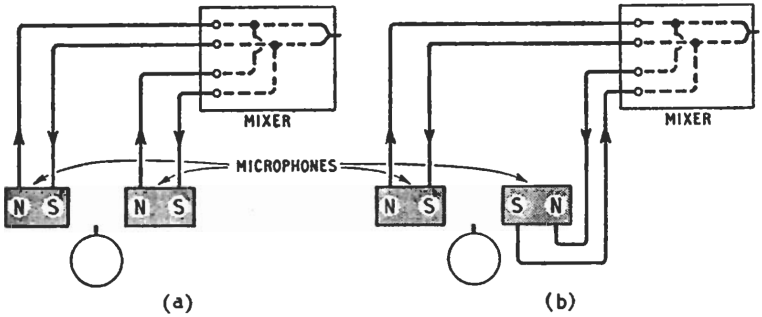


Fig. 13.1. Interference effect of two microphones equidistant from source—(a) in phase, (b) out of phase

be in step, as before, but currents of opposite sense will result due to the new positions of the poles of the magnet. In this case, the outputs of the two microphones tend to *cancel* at the input to the amplifier—the microphones are said to be " out of phase ". Under laboratory conditions, and when the microphones have identical response, the cancellation may be complete.

In practice, of course, such a positioning would be avoided, but whenever a layout approximating to this has been made necessary

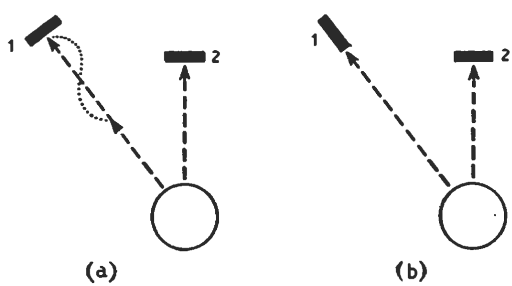


Fig. 13.2. General case of two microphones at different distances from scource (a) badly placed, (b) correct angling for minimum pick-up on Mic. 1

—e.g. in platform discussions—a reversing test should be carried out to find the better position.

13.1.2. General Case

If we now consider the case when different distances apply (Fig. 13.2), we see that the diaphragm movements are no longer in

phase, since the sound wave reaches the farther microphone (Mic. 1) slightly later. But there is one frequency (and harmonics) for which the extra distance corresponds to a wavelength and for which *addition* of the electrical outputs will take place. Similarly, another frequency (and harmonics) exists for which this extra distance is half-a-wavelength and *cancellation will result.*

This partial cancellation and addition of different frequencies amounts to distortion, and since special distances have not been considered, it follows that some distortion always results when two or more microphones (of any type) are faded up together.

In practice, it is usually possible, with all but omni-directional microphones, to reduce the distortion below the level at which it becomes serious by suitably angling the microphones so that *no two microphones pick up any source at anything like equal strength*—e.g. in Fig. 13.2(b), the source is placed effectively in the dead side of Microphone 1, and the contribution of Microphone 1 to the mixed output may be ignored (so far as source S is concerned). Using a close technique, as in multi-microphone dance band balances, will further reduce this trouble.

13.2. TALKS

Here we have the simplest type of balance problem, with one microphone placed centrally in front of a single speaker. The aim in a broadcast talk is to obtain faithful, pleasant transmission of

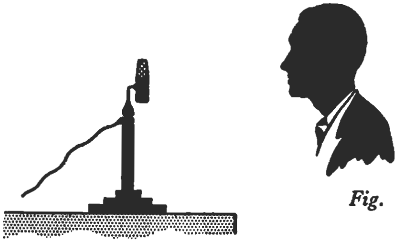


Fig. 13.3. Standard position for broadcast talk

the voice and to make it sound as it would in ordinary living-room acoustics. Given a good studio, the greatest single factor affecting the balance is distance from the microphone.

It is very important to avoid working too close, with the resultant exaggeration of bass frequencies. A minimum of 2 ft should be used, or 18 in. with a " corrected " microphone point (see Chapter 5). If a close technique is ever used for some special effect—such

as the " Mystery Voice " in " Twenty Questions "—it may be necessary to warn the speaker that distortion is likely on consonants like " p " and " b ". The trouble may be partially reduced if he speaks at a slight angle to the microphone.

Perhaps the most common acoustic defect in small studios is boominess due to resonances. It is often possible to reduce the distortion due to these resonances, and their accompanying pattern

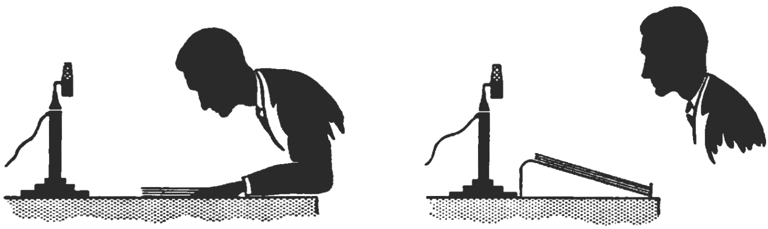
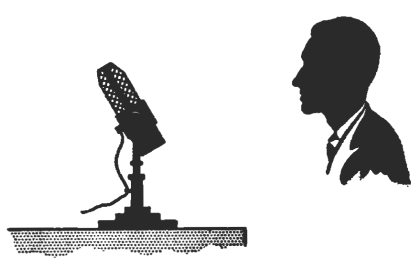


Fig. 13.4. Use of script-rest to correct looking down on to script

of standing waves, by exploring the available space. Working towards a corner may be tried, or placing the speaker near the centre of the studio, with the microphone slightly off-centre.

As far as possible, the speaker should be made comfortable, and allowed to take up any position at the table he likes (Fig. 13.3). However, positive action must be taken to discourage faults like speaking down on to the script, which causes distortion due to the interference of direct and reflected sounds, and the fact that more

Fig. 13.5. Tilting of AXB to reduce sibilance



high frequencies are reflected than low. A script-rest will help, as shown in Fig. 13.4. It is to prevent reflection of sounds into the microphone that studio tables are made with tops that are transparent to sound. Holding the script in front of the microphone is

another common fault, and results in selective masking of high frequencies, and further distortion.

When a speaker is noticeably sibilant, some improvement may result with a ribbon microphone if it is tilted as shown in Fig. 13.5. This will be more effective with older types of ribbon microphone having fairly large dimensions. In this case, attenuation of high frequencies will result since the live angle in the vertical plane is narrower for high frequencies (see Chapter 5). This tilting must be applied with discrimination to improve the overall pleasantness of the reproduction, without reducing intelligibility or changing the character of the voice.

13.3. INTERVIEWS AND DISCUSSIONS

Its double-sided nature makes the ribbon microphone very useful in studio interviews and discussions with up to four speakers. The AXB type ribbon is sometimes thought because of its size, to interfere with friendly discussion, and a dropped tray on elastic suspensions is let into some talks tables to allow the speakers to see each other more easily (Fig. 13.6).

When more than about four speakers are involved, a single ribbon microphone is often not sufficient. As Fig. 13.7 shows, the

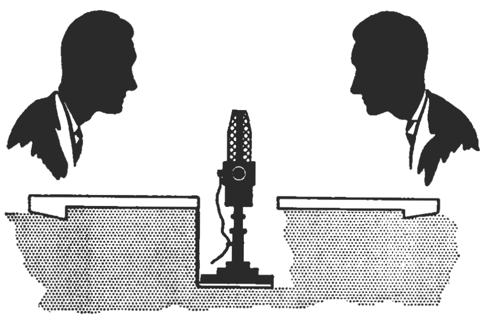


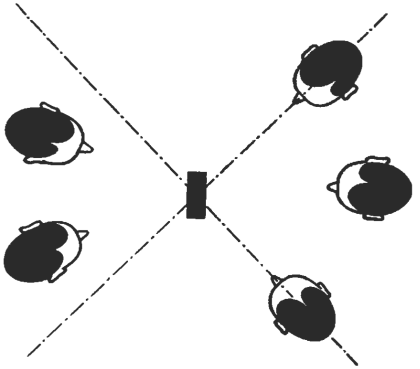
Fig. 13.6. Microphone let into well of talks table

live angle of 100° is too narrow to accommodate three heads at the normal working distance, and a greater distance may be impractical because of the studio acoustics. It is possible to get over this difficulty by using two ribbon microphones, fading them up

separately when a script is available. But the use of both microphones faded up together (unscripted programmes) requires careful handling, for the reasons explained in Section 13.1.

At the present time a cardioid microphone is usually placed with its working axis vertical below eye-level to give equal pick-up of all

Fig. 13.7. Discussion with five speakers, showing difficulty of using a single ribbon microphone.



speakers (or facing downwards from above eye-level). The Neumann KM54, A.K.G. C 12 and C 28 and the S.T. & C. 4033 microphones have all been used in this way with good results (see Plate 13.1).

The above remarks apply to " round table " discussions. If the speakers are placed " in line "—as for platform discussions—suitable angling of microphones (to avoid picking up any speaker on more than one microphone) will facilitate mixing. Sometimes a single cardioid microphone may be used on such programmes, when it must be set back so as to " see " the group as a whole.

13.4. DRAMATIC PRODUCTIONS

13.4.1. General

The types of acoustic for the various scenes in a studio production will be defined by the script, and each microphone should be carefully placed to get the desired effect—preferably before the cast assembles. Arranging for contrasting acoustics from scene to scene can often be very effective—a sharp contrast will frequently be preferred to actual realism.

Some years ago it was common practice to use several studios to obtain the desired changes of scene, carrying out the mixing in a drama control suite. This has the advantage of isolating the

scenes from one another, but was obviously prodigal of accommodation, and sometimes of manpower and rehearsal time. Wartime conditions largely put an end to this technique, and ushered in the General Purpose Studio, in which a " live " and " dead " end are provided, and several microphones are used.

Plate 13.2 shows Studio 6A in Broadcasting House. The microphone in the foreground was further deadened by the use of screens. In the " live " part of the studio a C.12 microphone has been set up to provide an alternative acoustic effect.

This type of studio is most successful when the " dead " end is really dead—i.e. carpeted, curtained off, and with absorbent treatment on all wall surfaces. Provided the treatment is sufficiently thorough, a series of usefully contrasting acoustics is possible.

13.4.2. Use of Studio Screens

All too often the need to accommodate other types of programme —music, etc.—means that the dead end is not made dead enough, and portable screens are necessary. These screens are helpful in two ways. Firstly, they increase the proportion of sounds reaching the microphone after reflection over *short* paths, and mask the microphone from *long* path reflections, so reducing apparent

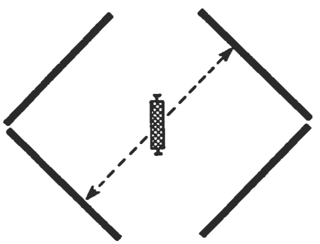


Fig. 13.8. "Tent" of screens, showing faulty position with screens parallel

reverberation. Secondly, by covering the surface of the screen with rock wool, etc., some absorption of sound is achieved, with further reduction in reverberation.

Unfortunately, portable screens suffer from the limitation that they are relatively ineffective at low frequencies. They are inefficient as reflectors for wavelengths over about 3 ft (frequencies below 350 c/s), and their absorbing efficiency also falls off rapidly in this region. Screens must therefore be used carefully if this frequency discrimination is to be kept to a minimum.

A " tent " of screens is often constructed, to obtain an open-air effect, i.e. no reverberation, as shown in Fig. 13.8. This requires

extra care, to avoid setting up parallel reflecting surfaces—with consequent distortion due to standing wave resonances, which will give a " box-like " acoustic. Superimposing records of bird-song etc., will not entirely mask the unrealistic effect of unsuitable studio acoustics.

13.4.3. Movements of Cast

Because of the microphone characteristics, the actual movements of the actors in a broadcast play will often require to be restricted—and not exactly what the context would seem to demand. Stepping back two paces, or into the dead side, may sound like retiring into the middle distance. When screens are being used, for example, exits should not be made into the open studio, but always into the non-reverberant part, so as to maintain the illusion of the particular setting.

Occasionally a scene can be acted more or less faithfully, in terms of movement, with excellent results. A court-room scene played in a large studio, with chairs laid out for judge, counsels, witnesses, public, etc., footsteps, and so on, may come to life over the microphone much better than if performed in a dead studio with artificial echo, and footsteps on a wooden board. The actors, too, will usually prefer the more natural conditions.

13.4.4. Overlapping Scenes

When using separate studios, it is a simple matter to superimpose scenes, but the microphones in a general purpose studio must be carefully positioned, especially if one of the scenes is a noisy one. The acoustics of both scenes are liable to suffer during the overlap, due to both microphones picking up the sound. If each sound source is arranged to be in the dead angle of all microphones except its own, mixing will usually be simplified.

A quick return to the studio after fading out, say, a crowd scene must be carefully rehearsed until everyone understands which light cue means " Quiet, please! ", and which " Start next scene ". Usually a series of flicks is used to obtain silence, and a steady light for the beginning of the new scene.

13.5. SINGERS

A minimum distance of about three feet from the microphone should be aimed at for solo singing. A much greater distance will be possible in some cases, depending on the relative loudness and

perspective of singer and accompaniment, and on whether the latter is piano, orchestra, or choir.

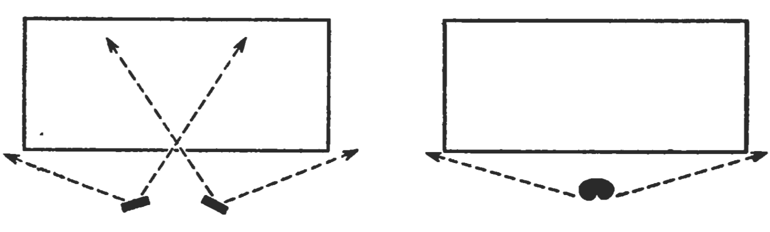
It is important to make the singers sound " in the same studio " as the accompaniment, and at a natural relative distance. It may be necessary to discourage undue movement while singing, in order to keep the conditions constant. The practice of " stepping back for the high notes " will sometimes give a disconcerting change in the apparent acoustic conditions. Certain experienced soloists have developed an efficient microphone technique, but generally it is better if they keep in the same position. If isolated notes tend to over-modulate, a *slight turning* of the singer's head might be suggested rather than stepping back.

Vocalists with modern dance orchestras will usually employ a closer technique, on account of the very loud accompaniment and (sometimes) the very quiet crooning. A Bass Correction Unit should generally be used, and explosive consonants watched out for as with close speech. Using a separate studio for the vocalist, and headphones to hear the accompaniment would simplify the separation problem, but it is not always practicable for other reasons.

13.6. CHOIRS

Choral works are generally most successful in fairly reverberant conditions. The blending of voices is improved, and indeed in some works there is little doubt that the composer has presupposed lively acoustics when deciding the overlapping of harmonies, etc. Moving the microphone further away—to get the best " blend "—tends to make the words less distinct, and the final position is usually at the greatest distance consistent with good diction.

With very large choirs, this limit may be reached at a distance from which the choir is not " seen as a whole " by a single ribbon



(a) (b)

Fig. 13.9. Large choir balance with (a) two ribbon microphones; (b) cardioid microphone

microphone, and two or three microphones may be necessary to preserve the internal balance of the choir. The microphones must be " phased " correctly and may be slightly angled to each other to minimise the area of overlap. A single cardioid microphone may be used for quite large choirs, and at a reasonable distance, because of its wide frontal angle of pick-up. It is sometimes necessary to discriminate in favour of the male voices when these are numerically in the minority (Fig. 13.9).

13.7. VOCAL GROUPS

Close harmony groups are a regular feature in Variety broadcasts, and present something of a problem, since they cannot be balanced satisfactorily on one side of a ribbon microphone. A split balance can give good results, but for audience shows a cardioid microphone is considered better from the presentation point of view. Larger groups, and small choirs, where a " tight " (close) balance is required, can sometimes be arranged in a V-shape on stepped rostra, with the leading voice in front.

13.8. SOLO INSTRUMENTS

13.8.1. General

As with singers, provided the increase in bass response at distances of less than 2 ft is avoided, microphone distance has little effect on the actual quality from musical instruments. The working distance chosen is therefore bound up simply with the acoustic effect required, and perspective and balance with respect to the other instruments present. (An exception exists in the case of instruments whose mechanical action is audible—celeste, guitar, piano accordion, and even violin—these suffer from too close a balance.)

Much more important, from the point of view of quality, is the *angle* the instrument makes to the microphone. As previously mentioned in connection with loudspeakers (Chapter 6), a source of sound waves is relatively non-directional at low frequencies, but begins to exhibit directional radiation as the frequency is raised to give wavelengths which are smaller than the source.

13.8.2. String Instruments

Violin

Except that radiation is more efficient from the front than the back, the violin is reasonably omni-directional at low frequencies.

At shorter wavelengths, however, the directional effect becomes more marked, and the microphone position becomes quite critical (Fig. 13.10).

If the higher harmonics are to be picked up, which give the characteristic attack and timbre, the microphone should be roughly

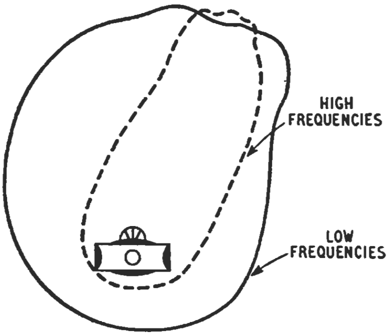


Fig. 13.10. Polar diagrams of violin, showing increased directivity at high frequencies

at right angles to the belly of the instrument. Note, however, that the reproduced quality will sometimes be considered better, and bow noise will be less, if the microphone is placed somewhat off this axis, even by as much as 45°. A compromise is inevitable.

Viola

Directional effects are less marked with the viola, and the microphone angle is not critical.

Violoncello

Excellent broadcast quality is obtainable from the 'cello, especially in the upper register. Because of the low playing position, however, it is important to avoid masking by other instruments. The directional properties of the 'cello are similar to those of the violin.

The soloist in a 'cello concerto is sometimes placed on a special rostrum. The resultant reinforcement of the sound may be an advantage but, of course, the rostrum may possess marked resonances.

Double Bass

The double bass is omni-directional over most of its lower register, and the principal balance difficulty is to ensure definition in this range. Floor and wall resonances tend to increase the blurred effect, and solid floors and walls are to be preferred. The

number of basses in an orchestra has long been known to be critical. One more or less than the optimum number seriously affects the balance of parts. In an orchestra large enough to employ four or more basses, it may be advantageous to arrange them on stepped rostra. They then present the appearance of an extended sound source and the resultant definition may be improved. The accuracy of the players' intonation is one of the most important factors in getting good definition.

Harp

The harp is not particularly directional in the horizontal plane, and the microphone angle is not critical. A balance combining good tone and suppression of mechanical sounds from a solo harp

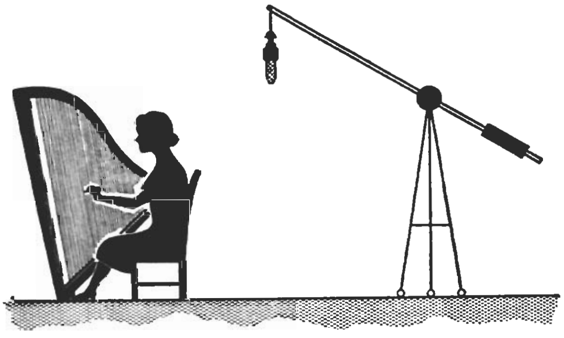


Fig. 13.11. A possible balance for solo harp

may be obtained with the microphone behind the player's head (see Fig. 13.11).

13.8.3. Pianoforte

As its name perhaps suggests, the pianoforte covers a wide range of dynamics. This, together with its percussive nature and richness in overtones makes this instrument a severe test of the technical efficiency of the microphone and the programme chain in general. The tonal qualities also depend on the studio acoustics and style of playing, to some considerable extent.

Depending on the circumstances, a good single microphone balance can usually be obtained within the arc shown in Fig. 13.12. The height, and angle of tilt of the microphone will affect this, and

at normal working heights it is common practice to work towards the tail end of the piano.

In some conditions there may be a tendency for the higher notes to recede in relation to the lower ones. In order to correct this error in sound perspective a second microphone can be used, placed higher than the lid of the piano, looking down on to the top strings. This microphone should be mixed in as required to restore the perspective balance.

The microphone distance depends on a number of factors, including the acoustics of the studio. For a piano recital of serious

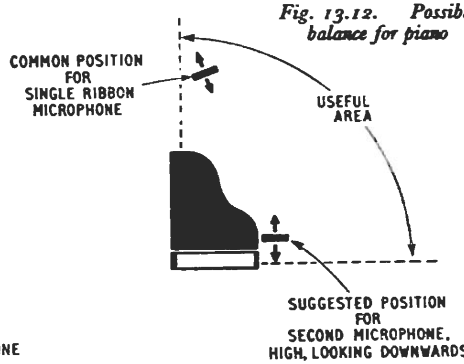


Fig. 13.12. Possible balance for piano

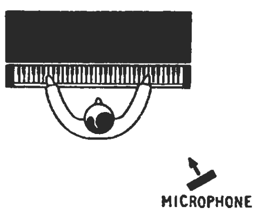


Fig. 13.13. Balance for upright piano

music, a distant technique is desirable, since a fair amount of reverberation is called for, and distance simplifies controlling—it being remembered that a Beethoven sonata, for example, may exploit the full dynamic range of the instrument. Thus, with a tilt of up to 45°, the tendency is to work as far away as the given studio will permit.

For light music, a less reverberant effect is often required, and a distance of about 4 ft is usual. Since the height will probably be reduced in proportion, care may be necessary to avoid the lid reflection.

The piano in modern dance orchestras is effectively part of the rhythm section. To cut out all reverberant sound and secure effective separation from brass, saxophones, etc., a very close balance is usual, with the microphone a few inches above the treble strings

(see Plate 13.3). It is common practice to remove the lid of the piano in order further to reduce unwanted reflections.

Upright pianos are not often used in broadcasts, but a balance which discriminates slightly in favour of treble is usually called for with the microphone in front as shown, or at an equivalent position at the back as in Fig. 13.13.

13.8.4. Early Keyboard Instruments

Solo broadcasts of the clavichord and instruments of the harpsichord family may present some difficulty. The extremely quiet sound of the clavichord dictates a close microphone position, but care must be taken to see it is not too close, or action noise will be troublesome. With the harpsichord, again, the microphone should be distant enough to avoid picking up the mechanical action. A programme volume corresponding to Peak Programme Meter readings of 3 or 4 maximum for the harpsichord and 2 for the clavichord, is to be expected, and unusually high control settings should be avoided in order to keep studio noise to a minimum.

Balance of the harpsichord with other instruments is often critical and depends to some extent on whether a figured bass form of accompaniment is being played or an actual melody line in polyphonic music. It is frequently difficult to obtain the correct balance with the right perspective relationships.

13.8.5. Woodwind Instruments

The flute, oboe, clarinet and bassoon require no special mention in this section, as their directional properties are not very marked. The microphone should not, however, be placed in direct line with the bell of the oboe or clarinet.

The french horn should be placed to face a given surface throughout in chamber or concert works, in case the player moves. In light orchestras, reinforcement of the horns can be achieved by backing them with hard screens.

13.8.6. Brass Instruments

Trumpets, trombones, etc., have considerable carrying power in a line from the bell of the instrument. The highest overtones are in fact confined within a narrow angle along this axis, and particularly brilliant quality is obtained from a microphone placed there (shared by any members of the audience who happen also to be in the " line of fire "). In solo work it is usual to place the microphone outside this narrow angle, the quality being less " hard ", and more

consistent, if the player moves slightly while playing. Exceptions are the trumpets in dance orchestras, and when a mute is being used.

13.8.7. Percussion Instruments

As mentioned in Chapter 2, percussion instruments can be divided into two groups; instruments with definite pitch, like the timpani (kettle drums), tubular bells, and glockenspiel; and instruments with no definite pitch, like the bass drum, cymbal and triangle. The sounds from this latter group are extremely complex, having component frequencies up to 20,000 c/s at least. Good results are therefore obtained only over the most efficient reproducing systems—and with very careful microphone placing.

The principal requirement here is definition, which would seem to call for a fairly close balance; but, of course, percussion must not be allowed to drown the other instruments. In practice, a compromise between definition and volume is usually reached with a single microphone, or a multi-microphone technique is used for good definition, and the musical balance is restored by mixing.

The notes of the *celeste* do not carry well, and a separate microphone may be necessary, except in serious works where the accompaniment is suitably thin.

The guitar in dance bands sometimes presents a twofold balance problem. At one stage it may be " acoustic ", when the radiation from the actual instrument is picked up; at another, it may be " electric ", when the player plugs in a loudspeaker, whose much louder tones are fed from an electric pick-up attached to the guitar.

13.8.8. Organ

A balance which reproduces the spacious effect of organ sound is preferred. The microphone should be distant enough for this to be obtained, but not so distant that the reverberation of the hall or studio masks the clarity of the instrument. Some exploring may be necessary to find the best position. In small churches, for example, the layout of organ pipes is not always symmetrical, a fact which may not reveal itself until a full listening test in the church has been carried out to locate pedal notes, etc. Similarly hidden attachments to cinema organs may exist such as tubular bells and electrically-operated pianos and drums, which the organist should be asked to demonstrate in case an extra microphone is required.

The small, transportable, electric organs often operate two loudspeakers. If this complicates the microphone placing, one of

these may be disconnected, or else a layout chosen which uses one loudspeaker for the audience, and the other for the microphone (Fig. 13.14).

Since an electronic organ generates electric currents to feed the loudspeakers, and the microphone converts the resultant sound back

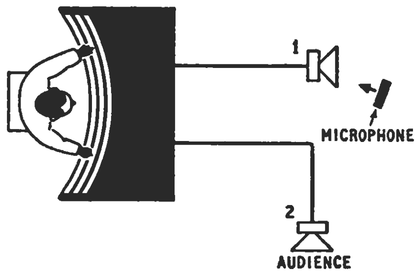


Fig. 13.14. Electronic organ, showing microphones placed to pick up one loudspeaker only

into electricity, it may be asked: " Why use a microphone at all? ". The alternative of simply connecting part of the organ's output to line has been employed in certain cases, but, of course, the absence of reverberation completely destroys the illusion of space.

The *piano accordion* should not be placed too near the microphone, because of key and reed noises, and usually the treble—i.e. the keyboard—end is arranged to face the microphone.

13.9. GROUPS OF SINGERS AND INSTRUMENTS

13.9.1. General

When several performers are to be grouped with respect to the microphone, every attempt should be made to place them comfortably in their natural formation. In the rare cases where an abnormal layout is called for, ease of performance must not be disregarded, nor the need for the musicians to see and hear each other easily.

In dramatic productions, distance from the microphone is adjusted to regulate the perspective of actors. Perspective is important in music, too—instruments should sound at natural relative distances—but the additional factor of balance applies in this case—viz. are we listening to melody and accompaniment, or melody and counter-melody, etc.? A complete understanding o the relative importance of the musical parts is essential in any studio manager who is attempting microphone balance of musical

programmes. This of course implies the ability to read a full score and discuss with the conductor and artistes any particular points of balance and interpretation to make sure that these are properly reproduced.

The choice of the type and number of microphones will be influenced by the kind of music—serious or light, etc.—and frequently by the necessity of producing particular effects in a given hall or studio.

13.9.2. Songs with Piano

It may seem a minor point, but it must be decided at the outset whether the piano is merely an accompaniment, or is of equal importance—as in the art song (*lieder*). It is safe to assume that the accompanist will maintain the appropriate level, but the position of the microphone must be right, in order that the appropriate perspectives should be maintained.

In the drawing, Fig. 13.15, position (a) has the advantage that the piano is not masked by the singer as at (b). However, this

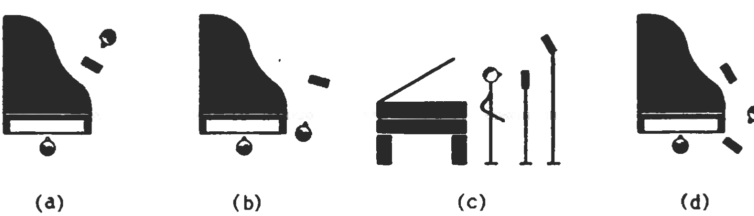


Fig. 13.15. Singer with piano accompaniment

masking effect is slight, and (b) allows much more freedom of choice for the microphone position—both in terms of height and distance, as shown in diagram (c).

A two-microphone balance which permits very close liaison between singer and accompanist is illustrated at (d). Note that the rule of non-pick-up of unwanted sources (Chapter 5) is strictly obeyed, and the dead sides of piano and singer microphones are directed towards singer and piano respectively.

13.9.3. Solo Instruments with Piano

Sonatas for violin and piano, etc., are usually balanced as at (b) in the preceding section, the soloist being placed at the end of the keyboard, and full advantage taken of the available space for

microphone placing. The novice is warned here, as elsewhere, against close balance. A distance and height of 9 ft for the microphone is not unusual.

13.9.4. Songs at the Piano

This balance is something of a compromise. A straightforward vocal balance is impossible since the piano would then be too loud,



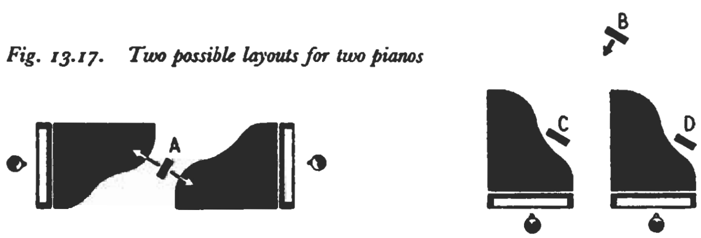
Fig. 13.16. Songs at the piano

and discriminating against the piano means that its quality suffers. In practice, the piano lid is put on a short stick, and the microphone tilted, as shown in Fig. 13.16, to reduce piano volume. A separate microphone may be mixed in to achieve better piano balance.

13.9.5. Two Pianos

Two common arrangements are illustrated in Fig. 13.17, and it will be seen that a single microphone is used. The position B is usually preferred, as greater liaison between the pianists is possible.

Fig. 13.17. Two possible layouts for two pianos



If the use of a single microphone gives too reverberant a sound—say, in light music—two may be tried as at C and D. With this layout, the lid of one piano is sometimes removed completely, but there seems to be little justification for this.

13.9.6. Chamber Music

A moderately live acoustic is nowadays aimed at for broadcasts of chamber music. It is imperative that no shift in relative perspective should take place during playing, and so a single microphone

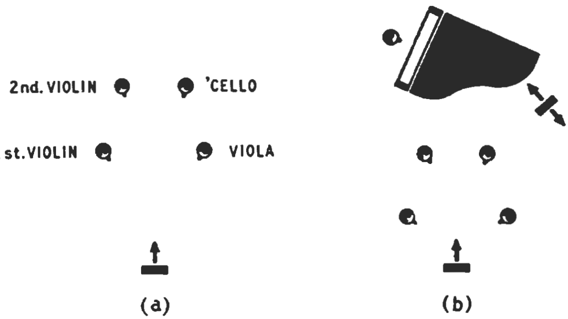


Fig. 13.18. Chamber music—(a) string quartet, and (b) quintet

is used, at a fair distance, and the players are grouped for ease of playing and equal balance (Fig. 13.18).

If a more intimate effect is required it may be necessary to use more than one microphone in order to preserve the relative perspective and balance between instruments.

If a piano or harpsichord is included, a second microphone (at a fixed level) may occasionally be required, as shown in the diagram. Variation of the " mix " of microphones is hardly ever necessary in chamber music.

13.9.7. Light Music Ensembles

A single microphone may well suffice for light music groups, plus a piano microphone if required.

The leader (first violin) of such a combination may remain standing, as shown in Plate 13.4, and may require an extra microphone for quiet solos. Modern arrangements, however, are tending to need a multi-microphone technique and two such typical layouts are shown in Fig. 13.19.

For larger light orchestras, modern scoring often makes a multi-microphone balance obligatory. The wide variation in the relative loudnesses demanded from the instruments in modern orchestrations may be reproduced only by the mixing of perhaps four or more

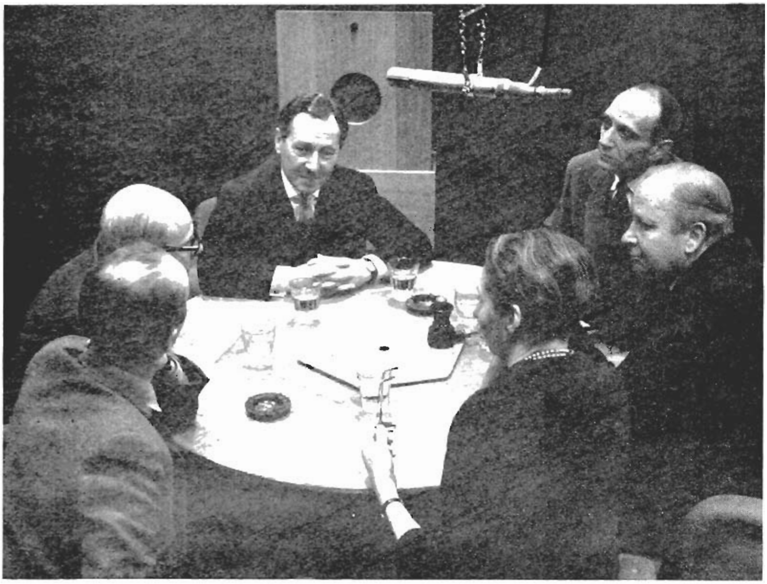


Plate 13.1. Discussion with five speakers, showing use of C.12 microphone slung over table

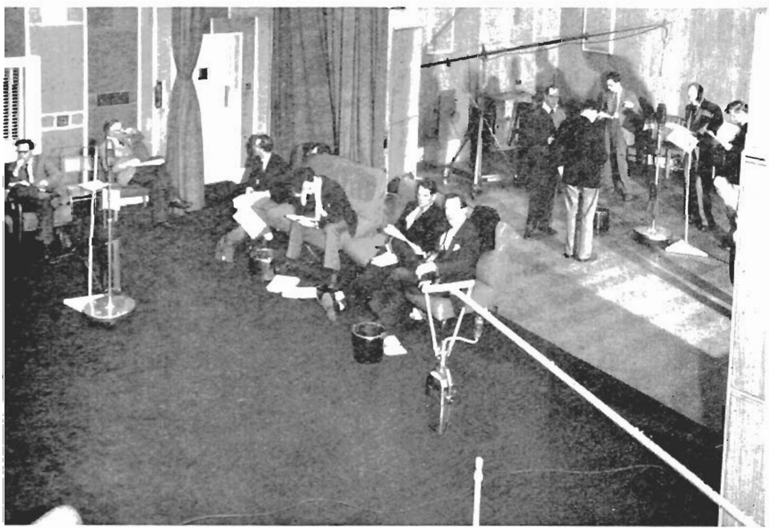


Plate 13.2. General view of Studio 6A, Broadcasting House, showing "live" and "dead" ends



Plate 13.3. Close piano balance



Plate 13.4. Small light music combination



Plate 13.5. Multi-microphone light music balance



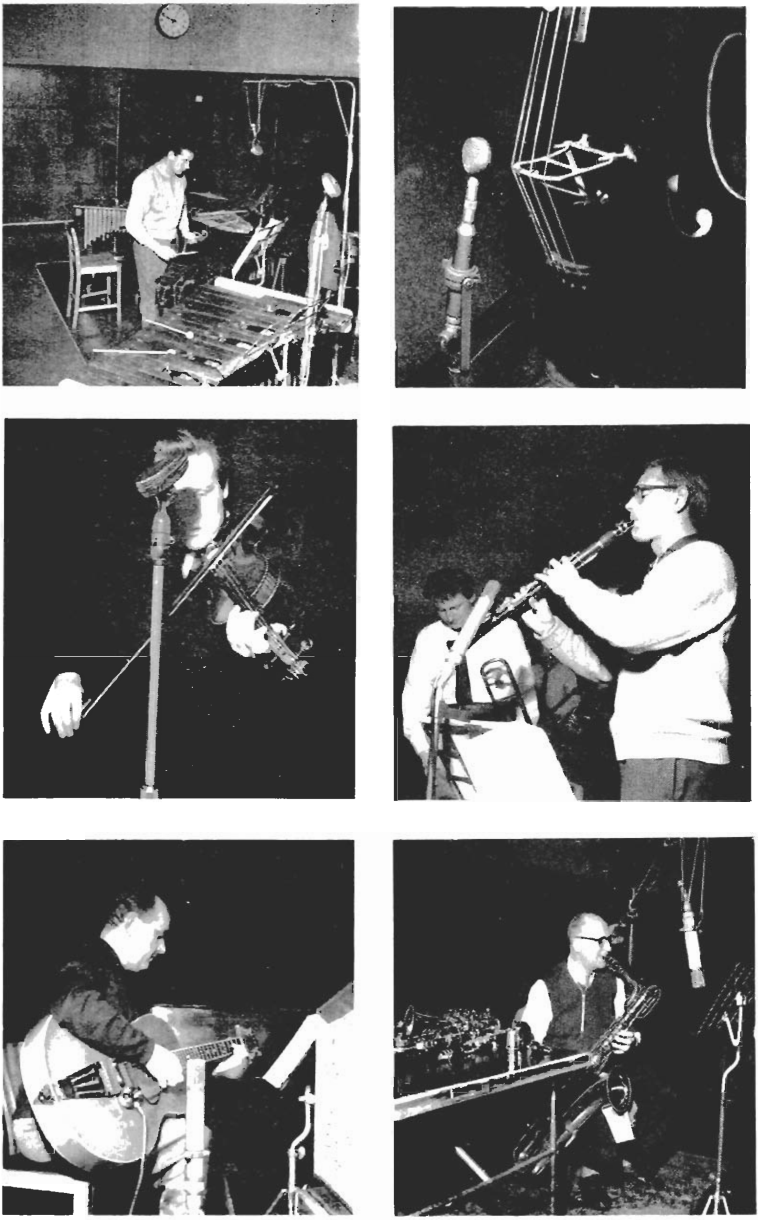
Plate 13.6. Use of a single C.12 microphone for the Royal Philharmonic Orchestra in Maida Vale, Studio 1



Plate 13.7. The Diz Disley Quintet



Typical microphone placings



for dance band instruments



Plate 13.8. BBC Variety Orchestra

microphones. The layout of the BBC Concert Orchestra in the Camden Theatre is shown in Fig. 13.20.

Plate 13.5 shows a complicated microphone balance in which the number of microphones exceeds the number of musicians.

13.9.8. The Classical Orchestra

The orchestra used up to about the year 1800 (Haydn, Mozart, etc.) is made up roughly as follows:

8 first violins, 6 second violins, 4 violas, 2 'cellos, 2 basses, 2 flutes, 2 oboes, 2 clarinets, 2 bassoons, 2 horns, 2 trumpets and timpani.

The layout of the orchestra—strings in front, then woodwind, and brass and percussion behind—is logical, since the carrying power of the instruments increases in that order. Also, in terms of the ratio of direct sound to reverberant sound, this layout gives best results. The attack and clarity of the string tone is best reproduced if direct sound is strong, whereas the blending of the more sustained brass tone is improved by reverberant conditions.

At least three possible seating arrangements of the strings are available to the conductor, reading from left to right:—

(a) 1st violins, violas, 'cellos, 2nd violins

(b) 1st violins, 2nd violins, violas, 'cellos

(c) 1st violins, 2nd violins, 'cellos, violas

From the microphone point of view (b) and (c) are perhaps to be preferred, since the violin sections are together.*

A single microphone should be used, placed to reproduce faithfully the internal balance of the orchestra under moderately reverberant conditions (Fig. 13.21). In cases where slight discrimination in favour of the upper strings is desirable—perhaps because they are few in number—a microphone at A or C will be chosen, and not B. Assuming a normal angle of tilt, A will help by its nearness to the upper strings, and C by its orientation. C will give better discrimination against trumpets and percussion in the layout shown, should there be any tendency for them to sound too loud.

The microphone distance is of primary importance. It must be far enough away to " see " the orchestra as a whole, and reproduce the internal balance, but not so distant that reflected sounds blurr the orchestral tone or articulation. In practice, depending on the

* For stereophony (a) may be preferred, in order to achieve the antiphonal effects desired by some composers.

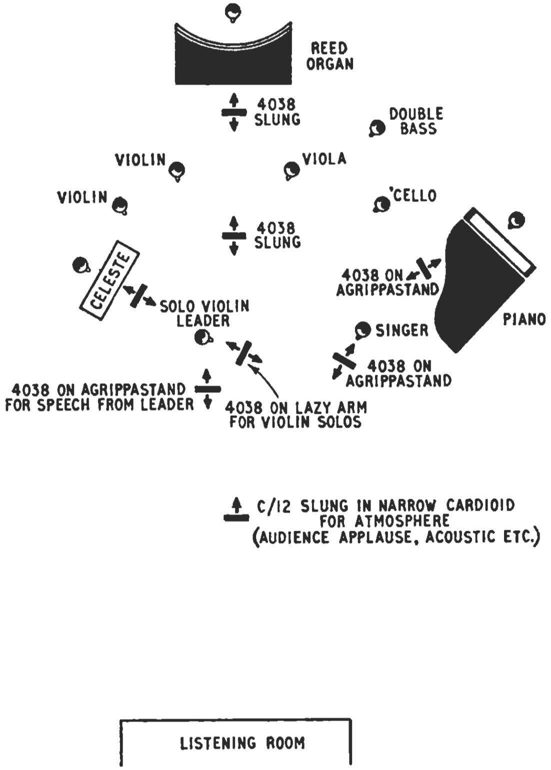


Fig. 13.19. (a) Microphone layout of "Grand Hotel"

hall or studio, distances from about 12 ft to 30 ft are used with ribbon microphones and slightly less with cardioid.

13.9.9. The Modern Symphony Orchestra

From Beethoven (1770–1827) onwards the orchestra has grown in size and variety of tonal colour. The modern composer makes use of an orchestra roughly as follows:—

20 first violins, 16 second violins, 12 violas, 10 'cellos, 8 basses, 3 of each of the following—flutes, oboes, clarinets and bassoons, with the third player doubling on piccolo, cor anglais, bass clarinet, and contra-bassoon respectively, 4 horns, 3 trumpets, 3 trombones, 1 tuba, 2 harps, celeste and 3 percussion players with a variety of instruments.

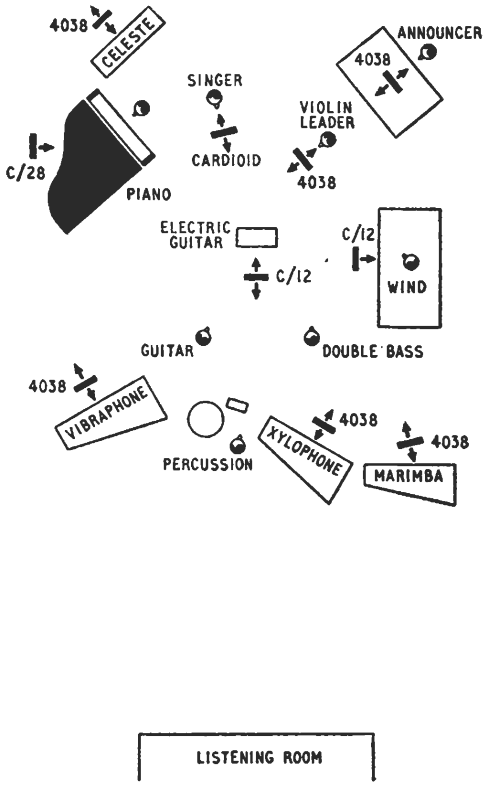


Fig. 13.19. (b) Microphone layout of "Charlie Katz Novelty Sextet"

All the remarks in the previous section apply here—with the added difficulty of the large area taken up by the players. The brass section, for example, may be as much as 30 ft from the front of the orchestra.

If a microphone were placed closer than 30 ft from the strings, the perspective (i.e. ratio of direct and indirect sound) would vary seriously throughout the orchestra. In circumstances where a distant microphone position is possible, the " out of focus " effect is overcome, as in Plate 13.6.

It must not be supposed that all sections of an orchestra should sound in the same perspective. In the studio, live, the brass and woodwind in fact sound more distant than the strings. This

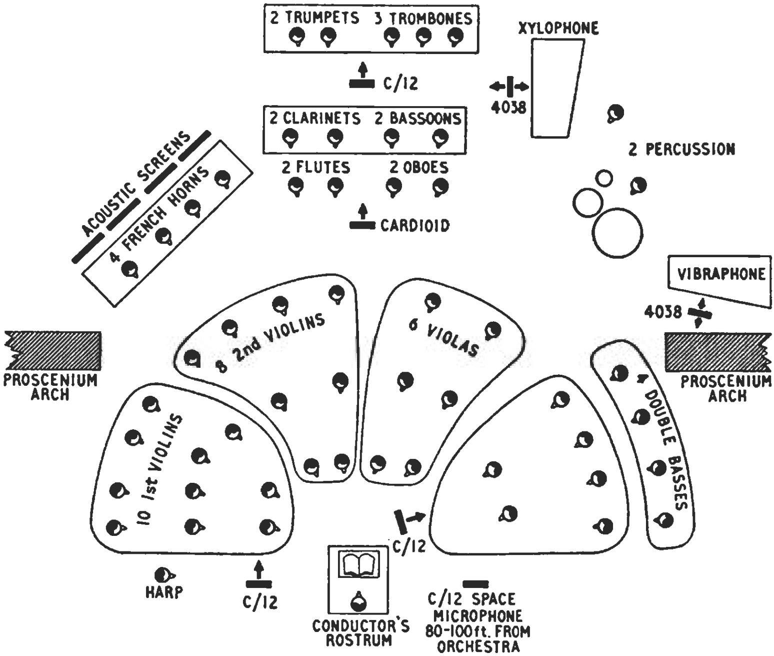


Fig. 13.20. Microphone layout of BBC Concert Orchestra in the Camden Theatre, London

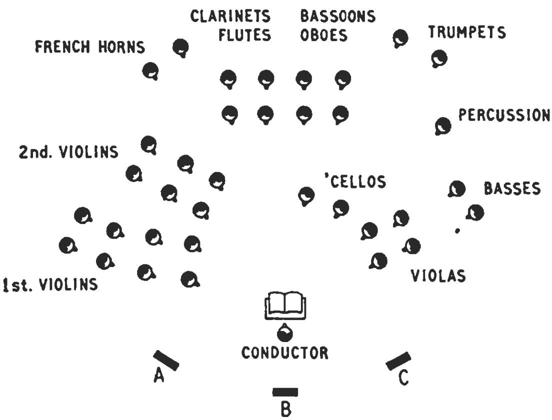


Fig. 13.21. A typical arrangement of the classical orchestra, showing three alternative microphone positions—A, B and C

natural effect of depth must be preserved over the microphone, in order to give " depth " to the orchestra.

13.9.10. The Concerto

Again, it is the true balance which is to be reproduced, and a single microphone should be used when possible.

A solo piano will usually be placed in front of the conductor, or slightly to his left, and other soloists—violin, horn, clarinet, etc.—at position X, in Fig. 13.22(a).

In cases where the concerto microphone is in a different position from that used for orchestra alone, it is very important to ensure that the orchestral perspective and the apparent acoustics are substantially the same on either microphone. The listener should hear

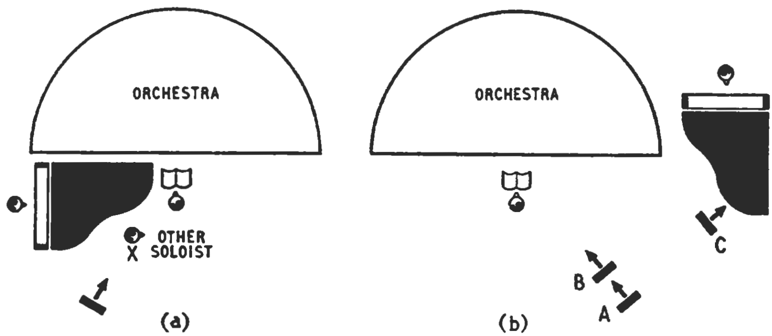


Fig. 13.22. The concerto—(a) normal arrangement, using single microphone, and (b) two-microphone balance

the *same* orchestra playing in the *same* place in orchestral and concerto items. When some form of compromise has been necessary, due perhaps to lack of space, the change-over of microphones should take place as imperceptibly as possible, or it may be preferable to take the concerto on the orchestral microphone, aided by very little of the output of a close microphone, as for singers in Section 13.8.11.

In Fig. 13.22(b), the piano has been placed to the side, and two microphones (B and C) are needed for the concerto. B is closer than A, which is used for orchestral items, since the added orchestral reverberation picked up by the piano microphone C must be compensated for. If the concerto, following an overture, symphony, etc., has an extended orchestral introduction (as in many classical

concertos), it may be preferable to remain on Microphone A for this, and change over to B plus C just before the entry of the soloist. The change in musical colour at this point will help to disguise the change in orchestral balance.

Needless to say, the above device may be unnecessary if applause and a long announcement separate the items, and will be impossible if the orchestral introduction is very short. The settings of faders B and C should, of course, be pre-arranged at rehearsal, and maintained during the whole work, as no shift in apparent perspective must take place.

13.9.11. Songs with Orchestra

In a concert hall, singers will normally take up a position at the front of the platform, and a subsidiary lower microphone will often

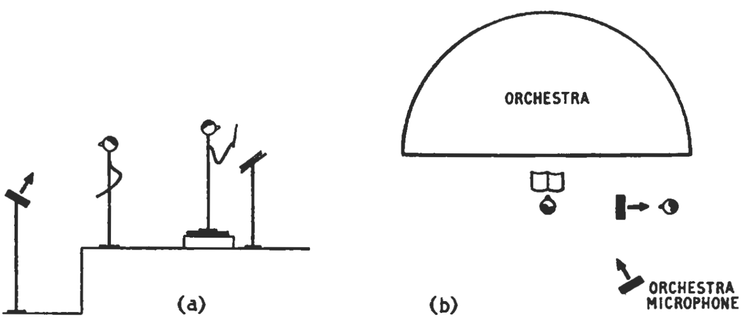


Fig. 13.23. Songs with orchestra. (a) shows one method of tilting the soloist's microphone, and (b) shows the normal studio balance with the singer's microphone at right angles to the orchestra

be necessary. Careful tilting will help to minimise perspective distortion on orchestral pick-up, as in Fig. 13.23(a).

In studio performances, singers may be placed on the conductor's left or right, as preferred, and the orchestral and vocal microphones may then be angled for easy mixing, in accordance with the principles set out at the beginning of this chapter (Fig. 13.23(b)).

13.9.12. Choirs with Orchestra

It is rare that choir and orchestra can be picked up satisfactorily on one microphone. The limiting factors are intelligibility from the choir, and definition from the orchestra.

In studio performances, when the numbers are not too great, the choir can be arranged at right angles to the orchestra—on the

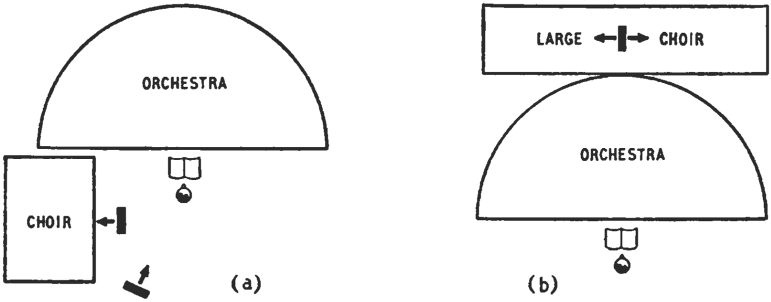


Fig. 13.24. Choirs with orchestra. (a) shows normal studio balance, and (b) shows split balance on large choir behind orchestra

conductor's left or on his right—with a separate microphone (see Fig. 13.24(a).)

When the choir is placed behind the orchestra, care is necessary to avoid pick-up of parts of the orchestra. The angle of tilt of the choir microphone is quite critical in this case. Very large choirs of 200 voices or more may make several choir microphones necessary, and diction is likely to be indistinct.

One or more ribbon microphones angled as shown in Fig. 13.24(b) have often been used successfully with large choirs. The use of cardioid microphones for choirs has already been mentioned above.

13.9.13. Brass Bands

The layouts for brass band broadcasts will usually conform to the pattern shown—namely, first and second cornets playing towards

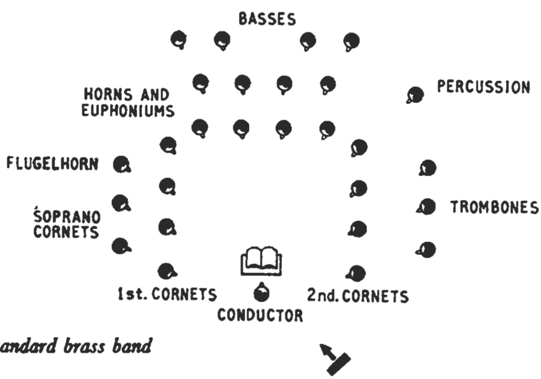


Fig. 13.25. Layout of standard brass band

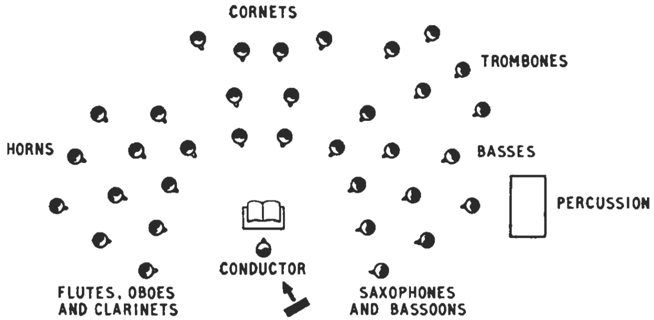


Fig. 13.26. Military band

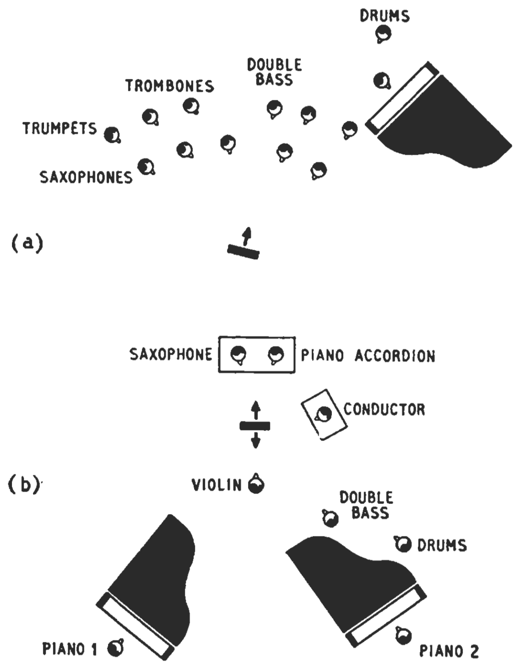


Fig. 13.27. Two examples of dance band balance with one microphone

each other, backed by sopranos and trombones respectively, with horns, euphoniums and basses making up the third side of a square (see Fig. 13.25).

This arrangement permits good definition on horns and euphoniums and prevents their being swamped by the direct tone of cornets and trombones. The basses, of which there are usually 2 E♭ and 2 B♭ instruments, are very important from the point of view of balance of the musical parts.

Cornet solos present no balance problem, but solos on the trombone or euphonium are a little difficult, because of the inherent

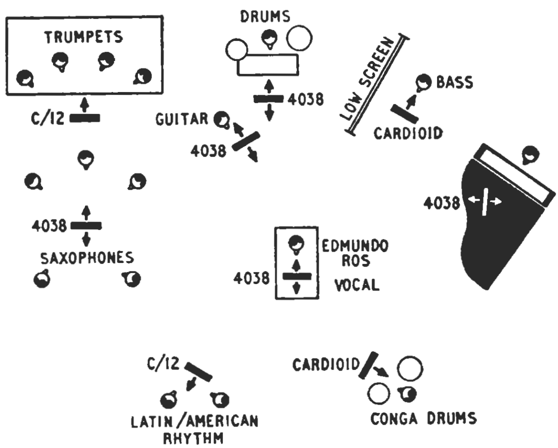


Fig. 13.28. Example of microphone arrangement for Edmundo Ros's band

weight of the accompaniment. In some cases a "spot" microphone may be used, or the soloists may be asked to come to the front of the band.

Artificial reverberation is sometimes used for brass bands when a close balance has been used for good definition, and natural reverberation is felt to be lacking.

13.9.14. Military Bands

A roughly three-sided layout is used again, with flutes, oboes, clarinets, and saxophones taking up the front rows. The percussion is more elaborate than the bass-drum, side-drum and triangle of the brass band; timpani and xylophone are quite common. Special attention should be given to the percussion effects at rehearsal, as precision of balance is required (Fig. 13.26).

13.9.15. Modern Dance Bands

Until a few years ago, a dance band played musical arrangements which were straightforward. The melody line was firm, and the accompaniment held at a level which presented a balanced sound to the listener on the spot. It followed that a one-microphone balance was possible, with occasional use of " spot " microphones.

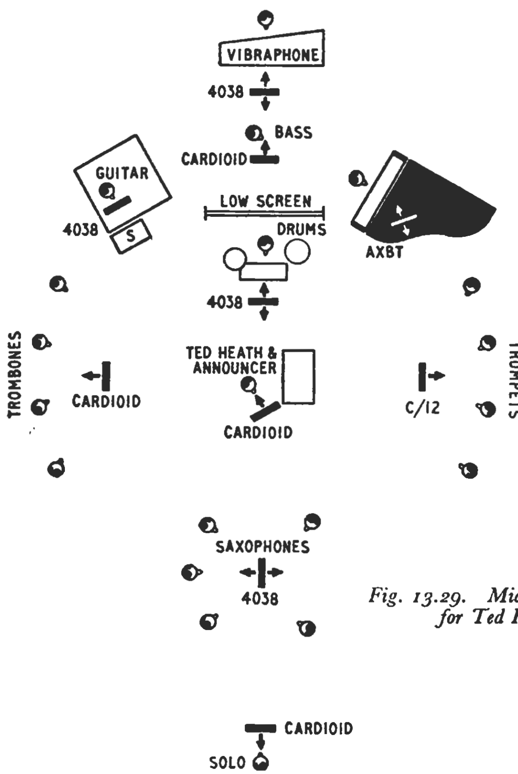


Fig. 13.29. Microphone arrangement for Ted Heath's band

Arrangements of this kind are occasionally met, and a layout based on Fig. 13.27(a) may be used. Another example of one microphone being used is the Victor Sylvester dance band, shown at (b).

When greater separation between sections of the band is required, and with the unnatural balances of up-to-date arrangements, e.g. celeste melody against a brass accompaniment etc., a multi-microphone balance is obligatory. A separate microphone is used for

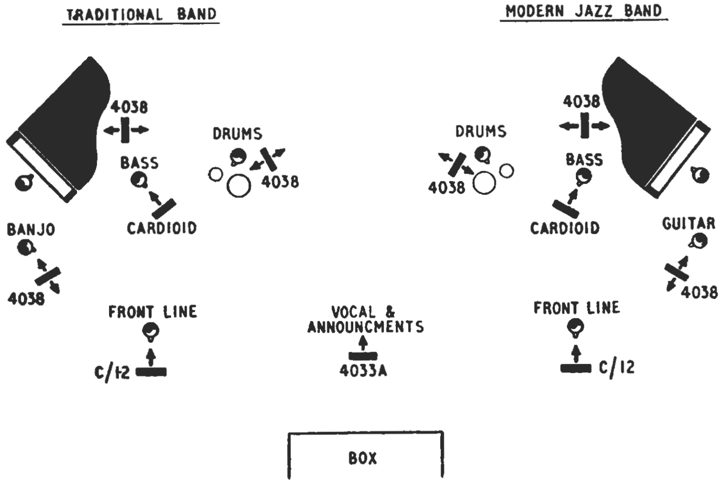


Fig. 13.30. Microphone arrangements for a traditional band and a modern jazz band

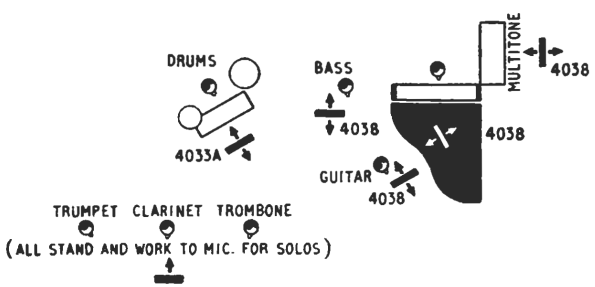


Fig. 13.31. Microphone arrangement for a jazz band

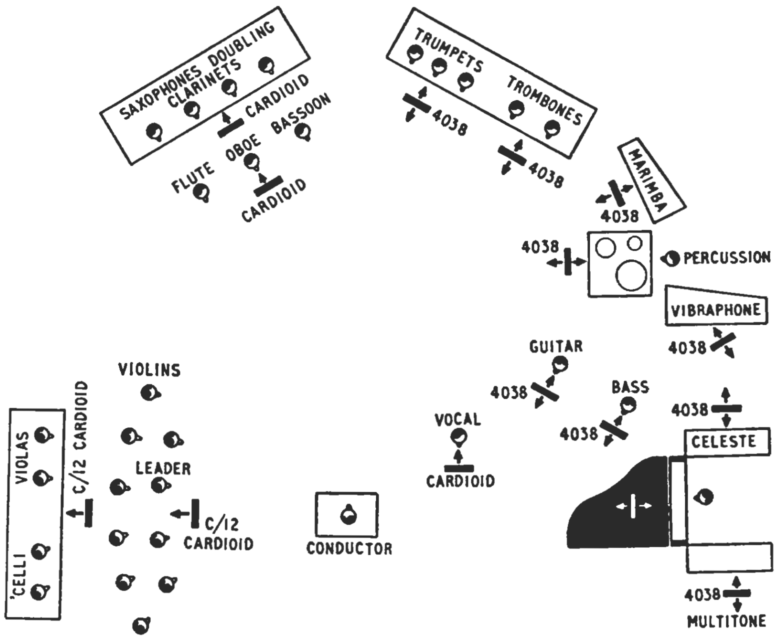


Fig. 13.32. Arrangement for a variety orchestra

each section of instruments, and careful angling, together with close positioning, serves to reduce out-of-phase effects to a minimum. Screens may be used to increase the separation between the various sections of instruments, and under some circumstances a " tent " of screens may be used for a vocalist.

Some typical arrangements of microphones used in jazz group and dance band balances are shown in Figs. 13.28 to 13.31 and Plate 13.7.

13.9.16. Variety Orchestras

The variety orchestra is a special case in that the music played may vary from the popular classical repertoire to typical dance band numbers, and so a balance technique capable of adjustment to these two conditions is essential. A typical layout is shown in Fig. 13.32 and Plate 13.8.

14

CONTROL OF PROGRAMME VOLUME

14.1. THE DYNAMIC RANGE

The volume limits between which satisfactory broadcasting or recording are possible in a given system define what is called the *dynamic range* of the system. In disc recording, for example, the groove spacing (dictated by economy of playing time) sets an upper limit to the recording amplitude; at the same time the lower limit will be fixed by considerations of surface noise. Between these limits we are normally left with a dynamic range of not much more than 30 dB.

In the programme chain for broadcasting, a number of other factors enter into the problem, such as transmitter modulation limits, level requirements in line circuits, etc., causing a further restriction of the dynamic range.

In the domestic services of the BBC, an average range of little less than 30 dB is aimed at (programme meter readings from 1 to 6, for example, correspond to a range of 22 dB).*

In the European and Overseas services, a narrower range is used, to allow for the extra noise and interference experienced on short-wave reception.

14.2. THE NEED FOR COMPRESSION

Compression is the name given to the process whereby the fluctuations in programme volume from a studio are restricted (automatically or by manual control) to conform with the narrower dynamic range of a given system. The diagram, Fig. 14.1, shows the approximate range of sounds generated in speech and in music. It will be seen that broadcast talks, which occupy a range of about 20 dB

* In primary service areas listeners, particularly on V.H.F., may well be able to use a dynamic range in excess of this. After all, there is an infinite number of decibels between 0 and 1 on the P.P.M. The restriction is only necessary when local noise is high.

may "escape" compression, whereas music (a range of approximately 60 dB) may require a measure of control to keep within the dynamic range of the system.

The drawing also shows the wider *frequency* range of music, compared with speech, which explains why attenuation distortion is more

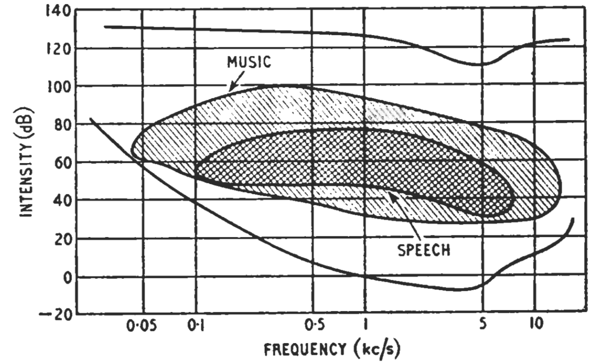


Fig. 14.1. Range of sounds in music and speech

serious (because it is likely to be more noticeable) on music than on speech (see Chapter 12).

14.2.1. Reason for Manual Control

It is possible to introduce compression automatically, using a limiter circuit, but results are usually disappointing from the artistic point of view. The musical expression, or light and shade, is a function of the fluctuations in programme volume, and in particular of the sharp contrasts in volume. These tend to be flattened out in automatic compression.

In manual control of the programme volume the studio manager or engineer is often able to anticipate contrasts in the level, and effect the necessary compression without seriously distorting the musical expression. He is assisted in thus imperceptibly controlling the volume by two characteristics of the human ear:—

(a) The ear cannot detect small changes in level (steps of 2 dB or less).

(b) The ear cannot compare accurately levels which are separated by a time lapse.

14.2.2. Procedure for Manual Control

Although there are nearly as many problems in controlling programme volume as there are types of programme, a rough procedure is outlined below as a general guide.

(a) Choose a studio layout and microphone position which will avoid, as far as possible, unnecessarily great contrast in volume (e.g. placing of soloists, effects microphone, etc.)—the aim being to reduce controlling to a minimum.

(b) Estimate early in the rehearsal the average setting of the main control potentiometer; all adjustments will then be made away from this setting, which is regarded as " home ".

(c) Note at rehearsal any points in the programme which exceed the permissible range with the control potentiometer at the " home " setting; and estimate the " away " setting that will be necessary.

(d) Anticipate these points by gradually fading up or down in advance (single stud adjustments at intervals of a few seconds will usually be imperceptible), so that the sharp contrasts are maintained, and not levelled out.

As an example, Fig. 14.2 shows a sudden peak followed by a gradual diminuendo. The control shown involves three down-

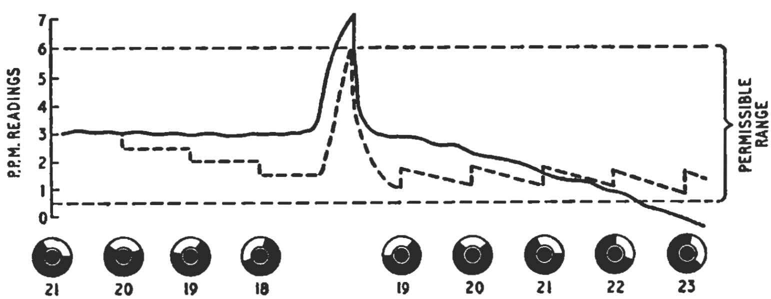


Fig. 14.2. Adjustments to control potentiometer in manual control of music levels

ward steps of 2 dB before the peak, and five upward steps during the quiet passage.

Although the above remarks apply most obviously to music programmes, they have a real application in dramatic productions also. Pistol shots, to take one of the many examples that spring to mind,

may be disappointing, not because of any real lack of pistol quality. It may simply be that they don't sound *loud enough*. If the preceding dialogue peaks 5, for example, and the explosion peaks 6, it will seem scarcely louder than the speech. Better practice would be to reduce the dialogue volume gradually in advance—say, to 3½—and then establish a contrast of about 10 dB. In the same way, contrast should be engineered between noisy crowd scenes and quite dialogues, etc., otherwise the crowd will not sound loud engouh.

14.2.3. The Limiter

However carefully manual control is carried out, there is always the risk that sudden peaks may exceed the permitted maximum. A *limiter* is therefore inserted into the programme chain before the transmitter or disc recording machine, as a means of protection against damage and distortion. The limiter takes the form of an amplifier whose gain is fixed for all normal levels of input. When the input exceeds the permitted maximum, the gain of the limiter is instantaneously reduced, so that a characteristic such as the dotted line " x " (see Fig. 14.3) is followed, and the output level cannot exceed 6. On isolated peaks, the recovery time of the limiter is very short, and the action is virtually free of the usual types of distortion.

The limiter, however, by its very action, produces abrupt distortion of the original dynamic range, and this can be very objectionable if the limiter is allowed to function on a number of successive peaks, or, indeed, over any long period of time.

14.3. PREFERRED VOLUME OF SPEECH AND MUSIC

We now come to the question, " Should music peak more than speech and, if so, by how much?"

The drawing in Fig. 14.1 indicates that music extends to higher intensities than speech—and we usually expect loud music to be louder than loud speech. As a general principle, therefore, it would seem natural to arrange that music peaks higher than speech. However, this is not always a desirable state of affairs; so much depends on the type of programme and the type of audience for which it is intended.

There are two basically different types of audience for a radio programme, firstly, the serious listener, that is the listener who does nothing else but listen to the programme, frequently on high quality equipment, and secondly, the listener who prefers his radio as a background whilst he is performing some other activity. Some

programmes may be considered as solely for one or the other, and suitable treatment can be arranged to cater for these. Other programmes may be listened to by both types of audience and in this case suitable treatment is more difficult.

In the first of the cases, the natural relationship referred to above, where loud music should sound louder than any accompanying speech, is usually preferred. Such a listener, for example, when

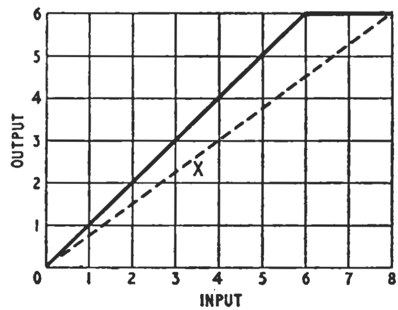


Fig. 14.3. Action of limiter

listening to an orchestral programme, will probably adjust his volume control to give a loudness approaching that which he would have heard in his favourite seat in the concert hall, and in this case he will require announcements within the concert, and, preferably the opening announcement to the programme following the concert, to be kept to a suitable low volume.

In the case of the background listener, he will adjust his loudspeaker to a fairly low level, such that the music is clearly audible but does not interfere with normal domestic conversation. Thus it will be by no means as loud as the previous case. If the announcements are then broadcast at a lower volume than the music, they may well fall below the conversational level and so will be inaudible. This will cause the listener to adjust his volume control frequently in order to keep the levels right for him, resulting in frustration and annoyance.

It seems quite clear that although the two programmes quoted above consist of speech and music, a different studio technique is necessary in so far as the adjustment of the relative levels of speech and music are concerned. It is the function of the production department to determine in which category a particular programme is placed and to instruct the studio staff accordingly.

It has been found convenient to apply the " background listening" conditions to many programmes taking place in the early morning,

and in some cases up to tea-time, and also to programmes in the early evening, when people are returning from work listening to their car radios, and housewives are preparing an evening meal. Programmes in the " serious listening " group will occur in the afternoon and, more particularly, during the evening transmission periods.

An account of investigations into listeners' preferences in this matter appeared in the BBC Quarterly for Spring 1950. The tests centred on the reactions of 60 people to a number of junctions between music and speech programmes at several relative levels. Programme junctions were used because most of the complaints from listeners about volume relate to the contrast between consecutive programme items. It is possible to summarise the results of the test in the following recommendations:—

1. Speech following music to be 4 dB down.
2. Music following speech to be 2 dB up.
3. Speech following speech to be within $\pm$ 2 dB.
4. " Bow Bells " interval signal to be 19 dB below speech.

From these recommendations the following points arise:—

(a) The discrepancy between 1 and 2 is not explained, but may be allowed for by starting a music programme 2 dB above speech, and then making up the other 2 dB when artistically possible in the music.

(b) Item 3 confirms the fact that intensity changes of 2 dB or less are imperceptible.

(c) Incidental music in drama and variety programmes is not, of course, covered by the above summary. In fact, there is considerable evidence that listeners prefer music within such programmes to peak several dB below speech. In dramatic productions, the music should never be higher, but certainly *lower* than if it were a concert item. Similarly, if the musical items are controlled to the same peak volume as speech in variety productions, they will often appear to be too loud.

14.4. PROGRAMME METER READINGS

It will have been noted in the above discussion that the *loudness* of the sound was referred to, rather than the P.P.M. reading. It is essential that for the purpose of varying such loudness, the studio manager relies upon his ears when listening to a loudspeaker adjusted

to a suitable level. It is possible that two sounds which to the ear sound vastly different in loudness, may give similar readings on the P.P.M., and so clearly this instrument cannot be used to give an indication of loudness. It does, of course, give an accurate indication of the peak volume permissible in order to avoid amplifier and transmitter overload and consequent distortion. Likewise, it gives a good indication of the lower transmission limits below which the volume should not go if the programme is not to be in danger of being swamped by noise.

The studio manager therefore, has a dual responsibility in this matter. Firstly, he must send to the listener the sequence of loudness which best represents the artistic content of the programme, bearing in mind the particular audience for which the programme is intended. Secondly, he must control the programme volume within the technical limits of the system.

15

SOUND EFFECTS

In many programmes, especially drama, operatic and light entertainment programmes, various sound effects are often required. These sound effects can be produced by three basic means:—" spot " effects, recorded effects and radiophonic effects.

15.1. " SPOT " EFFECTS

In the early days of broadcasting it was not practicable, because of the less advanced equipment, to produce convincing effect sounds on gramophone records. More often than not such a recorded effect would sound unconvincing, if only because the disc records of that time had rather high surface noise. Furthermore, and this problem remains at the present time, it was not possible to synchronise recorded sounds easily with live action in the studio. Because of this, a number of effects are actually created in the studio, and these are known as " spot " effects (Plates 15.1–3). They range from the more obvious sounds such as footsteps, doors opening and closing etc., all of which are the actual sounds of the respective pieces of equipment, to elaborate artificial effects, where the most unlikely objects may be pressed into service to create the required sounds. These last are seldom used in dramatic products at the present time, but still exist in some light entertainment programmes for comic effect.

For those interested in experimenting with unusual spot effects, a number are listed below:—

(a) Seawash: a handful of lead shot placed on the skin of a single-sided drum and gently rolled from side to side.

(b) Avalanche: a quantity of potatoes placed in a large bass drum and rolled from side to side.

(c) Squeak of car brakes: an inverted tumbler slid on a sheet of glass.

(d) Walking in snow: twisting a roll of new cotton-wool very close to the microphone.

(e) Machine gun: lead shot in a single-sided drum, vellum tapped with drum stick.

(f) Railway train: roller skate moved over stiff wire attached to board.

(g) Flight of arrows: swish of cane close to microphone.

(h) Creak of door: string attached to door handle, cloth impregnated with resin drawn along string as door is opened.

(i) Squeak of old inn sign: prongs of fork scraped to and fro round china plate.

(j) Windscreen of car splintering: wafer biscuit crushed close to microphone.

15.2. RECORDED EFFECTS

By far the majority of natural sound effects used in modern productions are specially recorded on tape, or more probably on disc. It has been found that 78 r.p.m. discs are most suitable for this work, since accurate location of any point on the recording is possible with suitable equipment. Fine-groove discs at 33⅓ r.p.m. can be used for sustained background sounds, such as crowd and restaurant noises, etc. Tape effects are used in many European countries and to some extent in the BBC, but they require a rather different technique in production. When discs are used for effects it is possible at any time during rehearsal immediately to change an effect for a more suitable one in the light of production experience. With tape, whilst this is obviously possible, more time is needed since the old effect will have to be removed from the composite effects tape and a new one inserted.

15.2.1. Recording of Sound Effects

Recording of sound effects is most usually achieved by means of battery-operated tape recorders, the tapes so obtained being later processed on to disc. The most important point to bear in mind when recording for sound effect purposes is that any extraneous noises present at the time of recording will, in all probability, sound quite incongruous in the situation for which the sound is intended, and indeed, even supposing such sounds were not out of place in a particular production the recording would almost certainly be kept for a later date, when such noise effects would be

quite unsuitable. In most cases when recording a sound effect, it is necessary to get as close to the sound as is conveniently possible, in order that background noises can be kept to a minimum. If the effect is required to sound at a distance in a given production, this can usually be achieved in the studio.

15.2.2. Reproduction of Recorded Effects

When recorded effects are to be reproduced into a programme, three methods are possible. Firstly, the output of the reproducing machine can be directly connected to the channel fader on the studio desk and the effect mixed in as required. This is the normal method of operation but can be a disadvantage under some conditions when it is necessary for artists to time their speeches to coincide with parts of the effect. In this case, two alternatives are possible. The recording can be reproduced, not directly into the programme chain, but on a loudspeaker in the studio, a system termed Acoustic Effects Reproduction (A.E.R.). By this means, the effect sound is audible to the actor in the studio and can be picked up by his microphone or another suitably placed. Thus relative perspective and timing between speech and effect can be controlled. The third method is a combination of the previous two, whereby the material is fed directly into the programme chain and also to the A.E.R. loudspeaker; the proportion can be varied to suit dramatic requirements.

15.3. RADIOPHONIC EFFECTS

With the development of the more imaginative type of radio production, it became necessary to invent sound effects which would suggest, in the imagination, particular emotional ideas rather than actual everyday sounds and to this end a new effects technique has been developed which has become known as Radiophonics. This term is not to be confused with the French " Radiophonique " which is merely an adjective applied to anything concerned with radio.

In the present sense, the result is a combination of the French " Musique Concrète ", the German and Italian electronic music, and straightforward sound effects. The sounds are created in the Radiophonic Workshop of the BBC by manipulation of tape recordings using a number of different methods (Plate 15.4).

15.3.1. Simple Manipulations

The simplest form of tape manipulation is probably that of changing the speed of the tape. If a natural sound, such as one

stroke of a bell, is recorded, and then the tape speed is reduced, a complete change in the character of the sound results. This is not only due to the reduction in pitch, but because the harmonic structure of the sound is altered by the speed reduction process. It is also possible, of course, to play such sounds in reverse, thereby converting the attack of the bell into an abrupt stop at the end of the sound. A sequence of such bell notes can be further altered in character, by suppressing the attack at the beginning of each note, either by means of a volume control or by physically removing it by

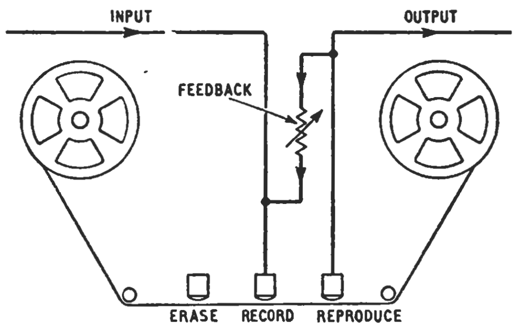


Fig. 15.1. Tape feedback

cutting it out of the tape. Obviously the number of sounds that can be treated in this way are limitless as also are the various adaptations of this simple treatment.

15.3.2. Echo and Feedback

Any of the methods of adding artificial echo or reverberation used in the studio can be added to the sounds made by tape manipulation. Further effects are possible by yet another use of the tape machine. If the replay amplifier output, on a given tape recording of the sound which is to be treated, is fed back via an attenuator to the record amplifier, the physical separation between the record and replay heads of the machine will produce a repetitive effect akin to an echo, the amplitude of which can be controlled up to oscillation point by means of the attenuator (Fig. 15.1). The delay can be further increased and extra echoes added by allowing the tape to pass from the first machine and across the replay heads of any number of other machines before being taken up on the last

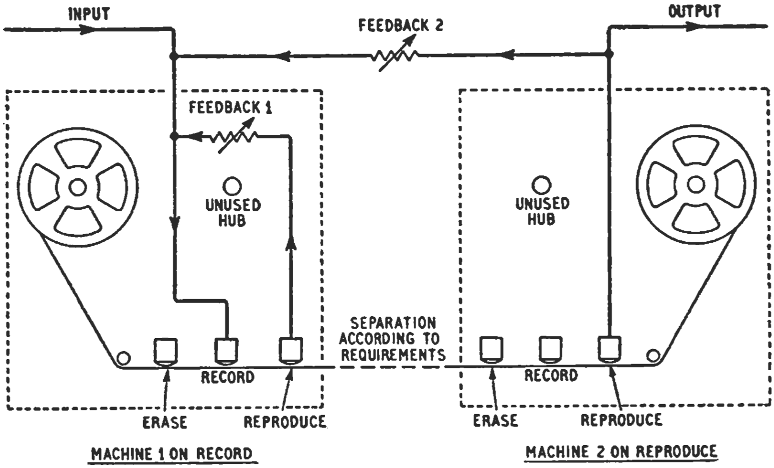


Fig. 15.2. Tape feedback—delayed

machine. The outputs of all replay heads in the chain can be fed back in varying amounts to the first record head (Fig. 15.2).

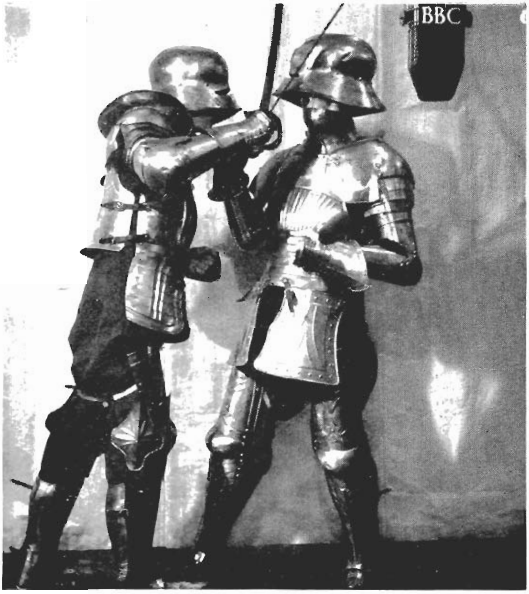
15.3.3. Repetitive Sounds

Having produced a number of basic sounds, by the means outlined above, these can be cut and joined into loops of tape, which when placed on a machine will play continuously for as long as required. A number of such loops on different machines can then be combined on a mixing desk to produce a " multi-layered " montage of sound.

15.3.4. Extra Equipment

So far all the equipment described has been of the basic variety, namely tape machines, tape and editing material. In addition to these, oscillators are required for the production of accurately determined sine wave tones (Plate 15.5). These tones can be combined in differing frequency and proportion, and constitute another source of basic sounds; they can then be added to the sounds produced from natural sound sources and manipulated in the same way. The pure sine wave need not be used; it can be passed through a square-wave shaping device which will produce a sound rich in harmonics. This can then be modified by means of filters to produce further different sound colours (Plate 15.6).

Plate 15.1. Studio Managers making sound effects for " Morte d'Arthur "



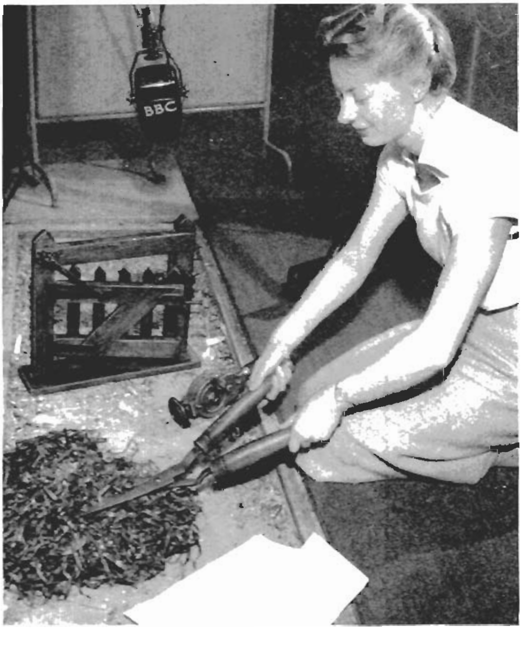


Plate 15.2. Studio Manager " cutting the grass " for a garden scene



Plate 15.3. The " spot " effects store





Plate 15.4. BBC Radiophonic Workshop

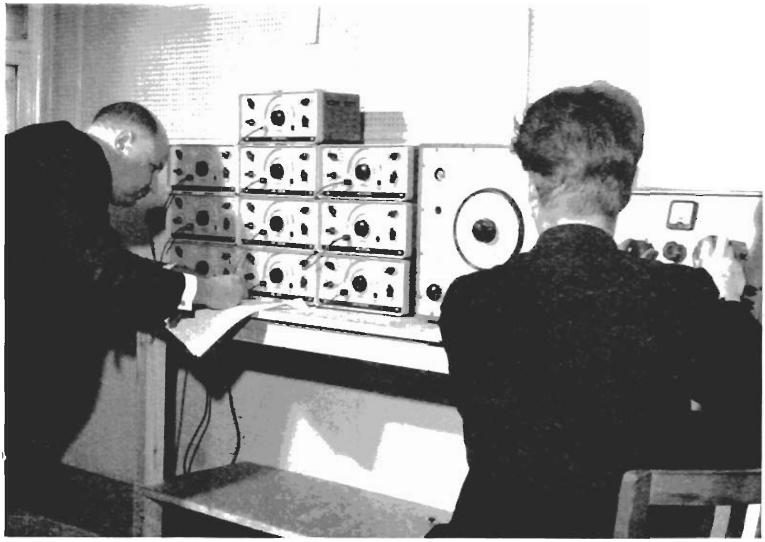


Plate 15.5. Bank of oscillators, showing use of frequency standard (on right) to set up accurate combinations of tones

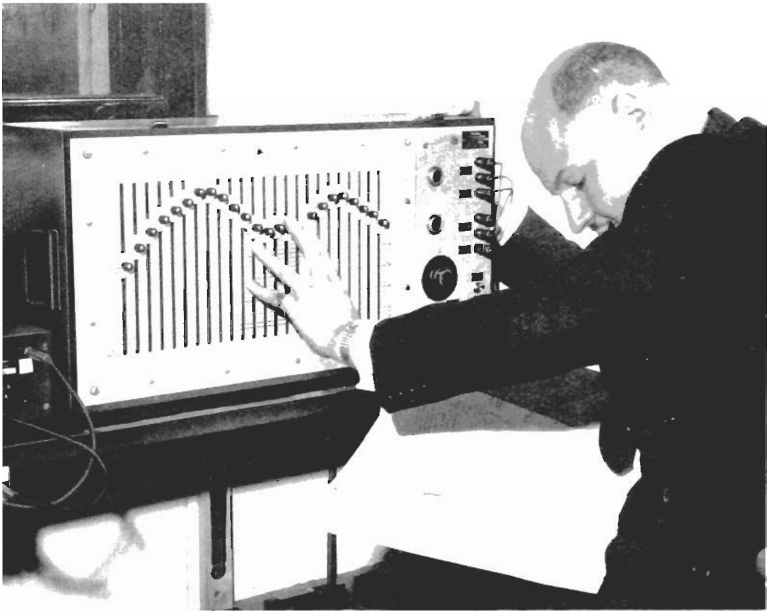


Plate 15.6. Filter, variable from 0–60 dB attenuation in ⅓ octave steps

Filters can be of several types; high pass, low pass or band pass, and can have varying degrees of attenuation.

Yet another source of basic sound used in radiophonics is the *white noise generator*, a device which produces equal intensity of sound over the whole of the audible frequency spectrum, a sound which if heard on a loudspeaker is akin to that of escaping steam. Since this sound contains all possible frequencies in equal amounts, sharp filtering can produce bands of noise which can have characteristic pitch, and these can be used in a similar manner as before.

The above description gives an outline of some of the processes of producing sound effects by radiophonic means. Obviously, the potentialities are limitless and new sources are continually being devised. The BBC's Radiophonic Workshop produces such effects for television and sound programmes and the demand in both directions is increasing. It is hoped that the brief description in this chapter will make it possible for others to attempt the creation of effects in this way.

One might have thought that such effects are new, but the following extract from " The New Atlantis " by Francis Bacon, written in 1624, seems to suggest it may all have been done before!

> " Wee have also Sound-houses, wher wee practise and demonstrate all Sounds, and their Generation. Wee have Harmonies which you have not, of Quarter-Sounds, and lesser Slides of Sounds. Diverse Instruments of Musick likewise to you unknowne, some sweeter than any you have; Together with Bells and Rings that are dainty and sweet. Wee represent Small Sounds as Great and Deepe; Likewise Great Sounds, Extenuate and Sharpe; Wee make diverse Tremblings and Warblings of Sounds, which in their Originall are Entire. Wee represent and imitate all Articulate Sounds and Letters, and the Voices and Notes of Beasts and Birds. Wee have certaine Helps, which sett to the Eare doe further the Hearing greatly. Wee have also diverse Strange and Artificiall Ecchos, Reflecting the Voice many times, and as it were Tossing it: And some that give back the Voice Lowder than it came, some Shriller, and some Deeper; Yea some rendering the Voice, Differing in the Letters or Articulate Sound, from that they receyve. Wee have also meanes to convey Sounds in Trunks and Pipes, in strange Lines, and Distances ".

16

STEREOPHONY

In recent years considerable interest has been aroused in the possibility of stereophonic reproduction under domestic conditions. So far, only a limited amount of broadcasting on an experimental basis has been possible owing to various transmission difficulties, but numerous recordings both on disc and tape are available for purchase on the commercial market. It is felt that this book would not be complete without some reference to the various systems of achieving a stereophonic sound picture and some description of equipment and studio technique designed for this. For the purposes of this chapter we shall consider only domestic stereophony, that is, two-channel stereophony, since it seems unlikely that any system using more channels than this will be economically or technically possible for some time. Multi-channel systems are used in cinema stereophony but these will not be discussed here.

16.1. HISTORY

It is probably fair to say that ever since the invention of the microphone and the headphones, and later still the loudspeaker, attempts have been made to achieve reproduction which would simulate the spatial distribution of sound normally heard in everyday life. The earliest recorded experiment was towards the end of the last century when two telephone-type microphones were placed in the footlights of the Paris Opera and their outputs separately connected to pairs of headphones. The report of the experiment said that listeners were able to locate the position of the performers on the stage. Many other experiments were conducted before an acceptable system of stereophony on loudspeakers was achieved, but two series of experiments, both carried out in the 1930s, can fairly be said to have pointed the way to present techniques. One of these was conducted by the Bell Telephone Laboratories in the U.S.A. and resulted in a number of demonstrations,

notably the transmission over land lines to Washington in 1934 of a concert performed in Philadelphia. Three years earlier, in England, A. D. Blumlein, of the Columbia Graphophone Co. Ltd., also conducted experiments culminating in the filing of British Patent No. 394325, which describes a complete system of stereophonic recording and reproduction from microphone to disc: this, although not practicable at that date, has become the basis of the modern stereophonic long-playing record.

16.2. DIRECTIONAL HEARING

Before discussing these two main systems in detail, we will consider how the human brain interprets sounds arriving at the ears in order to determine the position of those sounds. Early theories on the subject suggested that it was the difference in loudness at the two ears which gave the required information, the head having a shadowing effect on sounds arriving from one side. Lord Rayleigh, in 1896, in his " Theory of Sound ", pointed out that this intensity difference would only be significant at the higher frequencies, above 700–1,000 c/s, where the wavelength of the sound is shorter than the distance between the ears. He suggested that the low frequency directional information might come from the difference between the time of arrival of the sound at one ear and at the other.

More recent experiments on the subject, notably those by Cherry and Leakey at Imperial College, London, confirmed the low frequency suggestion but indicated that the effect at the high frequencies was also produced by time difference and not by the amplitude difference. In this case, however, it was not the difference in time of arrival of the high frequency waves themselves which gave the required information. In natural sounds, almost all high frequency components are varying at some lower frequency (e.g. the " warble " of bird song, the vibration a violinist imparts to his notes, and the beats between various sounds) and it was this " modulation envelope " which supplied the information to the brain. " Pure " high frequency sounds, tones, for example, were almost impossible to locate.

The experiments further showed that the effect of inter-channel time difference is to blur the sharpness of the reproduced image.

It would seem likely therefore, that a loudspeaker system of reproduction which could reproduce at the ears of the distant listener the interaural time difference that would have been heard by a listener in front of the original sound source, would achieve the

desired result. Both the Bell and Blumlein systems of stereophony attempt to achieve this.

16.2.1. "Wavefront" System

The Bell system of stereophony has been termed the "wavefront" system because its original idea was a reduction of an ideal situation where a "curtain of microphones", infinite in number, was

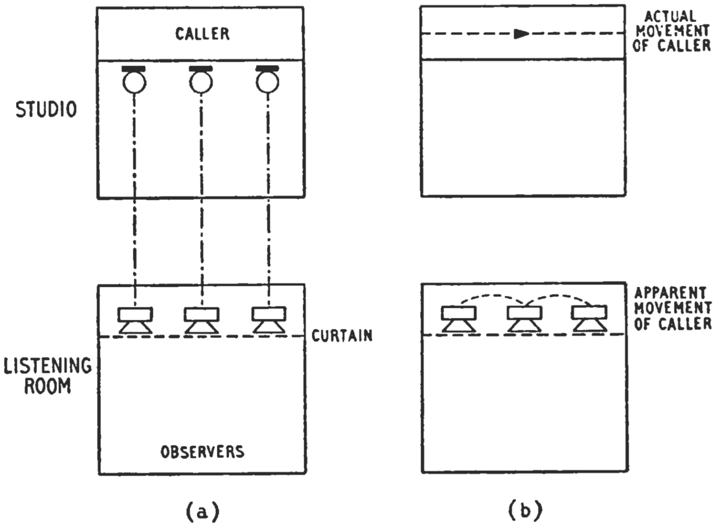


Fig. 16.1. Three spaced microphones—three loudspeakers

supposed to be hung in front of the sound source, each microphone being connected to a corresponding loudspeaker in a similar "curtain" in the listening room. In this imaginary case the whole wavefront of sound leaving the sound source, should be continued from the loudspeakers in the listening room. Obviously such a hypothetical "sound-curtain" would be quite impossible and so the Bell engineers reduced their number of microphones and loudspeakers to three and later to two. For the purposes of the experiment, a "caller" moved about to a number of pre-determined positions in front of three omni-directional microphones arranged in line in a studio. In a small auditorium three loudspeakers, one connected to each microphone in corresponding positions on the platform, reproduced his voice, and observers were required to mark

the apparent position of the " caller " on a plan (Fig. 16.1(a)). A sound-transparent curtain was hung in front of the loudspeakers to assist the illusion. To complete the experiment the " caller " made his moves behind this curtain in the listening room, no electronic equipment being used. It was found that fair accuracy of positioning across the stage was possible, but that the positions in between any two microphones produced an effect in the listening room of the " caller " moving upstage away from the observers, although he did not in fact do so. This might have been expected, since he was then farther from either microphone. If the " caller " walked from one side of the stage to the other in a straight line, it appeared from the listeners' point of view that he walked in a curve slightly upstage at first, coming down to the front again by the centre microphone and then up again and down towards the microphone at the other side (see Fig. 16.1(b)). If the centre microphone was removed, thus giving a two-channel system as domestically desirable, this effect was rather worse, there being a distinct recession of sounds in the centre of the stage. This simple system, using two omni-directional microphones, was the basis of many early stereophonic recordings and gave rise to the criticism that such stereo had a " hole in the middle ". An attempt to fill up this hole has been made by reintroducing the centre microphone and connecting its output not to a third loudspeaker, but equally to the other two,

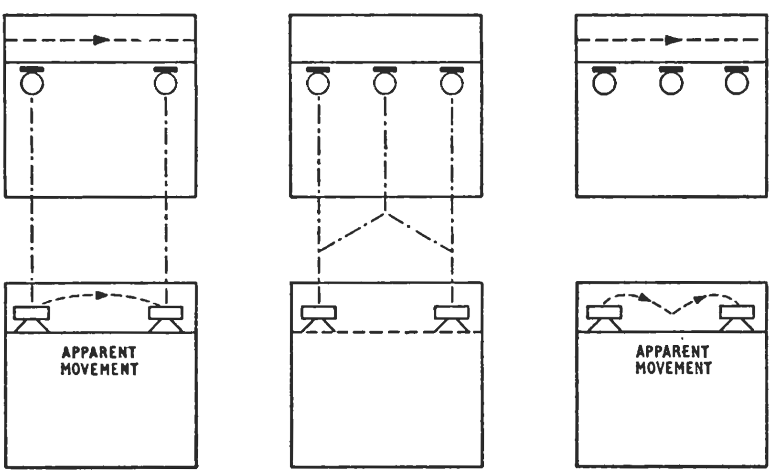


Fig. 16.2. Three spaced microphones—two loudspeakers

thereby producing equal outputs from the two loudspeakers, a condition that would give a centre image. This is more or less successful, but tends to produce two lesser " holes ", one-third and two-thirds the width of the stage, unless care is taken in microphone placing (Fig. 16.2).

Many American recordings made at the present time use this system as a basis although other reinforcing or " spot " microphones may be added to assist in the general effect. This system is capable of good spatial distribution of sound but may not give " pin-point " positioning of component parts of a complex sound source; for example, an oboe player in a symphony orchestra. It will, however, give good spatial representation to the various sections of an orchestra: for example, it will separate wood-wind from strings or brass.

16.2.2. Coincident Microphone System

To turn now to the Blumlein system. In addition to describing a spaced microphone system using omni-directional microphones, similar in many ways to the Bell system, Blumlein also described a system using two very closely placed directional microphones. This system has become known as the " coincident " microphone system, since ideally the microphones should in fact be in the same position. This, of course, is not possible and in practice they are mounted as close to one another as possible. The operation is as follows: if we refer back to the theories of directional hearing discussed above, we will see that in order to produce a similar effect in space to the original sound, we have to create in the ears of the listener the same time differences that he would have heard in a corresponding position in front of the original sound. It can be shown that in order to do this using two loudspeakers all that is necessary is to feed the loudspeakers with sounds in differing amplitudes, this amplitude difference being proportional to the angle at which the sound arrives at the studio microphone. Two figure-of-eight microphones mounted with their main axes at 90° will produce differing outputs depending on the angle of incidence of a given sound to each microphone. Such a microphone pair, therefore, will provide the necessary transformation between position in space, and amplitude difference between loudspeakers, to achieve the desired effect (Fig. 16.3). Microphones of other directional characteristics can also be used, depending on the type of pick-up required. In this case good positional accuracy is possible, limited by the acoustic conditions in the studio and more particularly in the listening room.

16.2.3. Choice of Systems

Which of the above two methods is to be used in a complete system of domestic stereophony is open to some discussion, but if such a system is required to transmit accurate movements of artists, say, in a dramatic production, that which provides more accurate

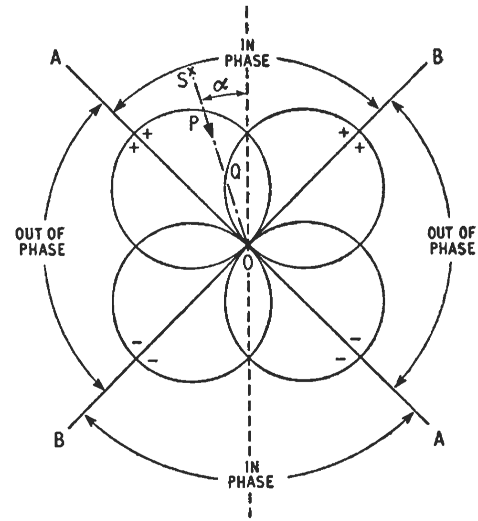


Fig. 16.3. Coincident figure-of-eight microphones at angle α, then output from Mic. A is represented by PO and from microphone B by QO

positional information is likely to be preferred. This therefore suggests that the coincident microphone system should be used as a basis. This, and its developments, will be discussed in the remainder of the chapter.

16.3. MICROPHONE TECHNIQUE

In many programmes, particularly serious music, a simple set-up employing one coincident microphone pair will be all that is necessary. For example, a symphony orchestra without a soloist, and in some cases with a soloist, will usually be quite satisfactory with this technique. Similarly, a small chamber group is also a simple matter to reproduce in this way. Difficulties begin to arise, however, when, for example, the orchestra has a solo instrument and the scoring is such that the solo line will tend to be obscured by the orchestra. In monophonic balance technique, it would be simple to add a second microphone close to the soloist and to fade this up sufficiently to restore the desired balance. In stereophony, however, although this can be done, some consideration of the

consequences of using various types of microphone is necessary so that the required result can be obtained.

We have seen that our main stereophonic microphone pair can consist of two directional microphones with their axes at 90°; in this case the acceptance angle of the " stereo microphone " might be expected to be 90°. In fact the angle with figure-of-eight microphones set at 90° is rather less than this, about 80°, and with cardioids rather more, as much as 160°. Any sound which lies on the left-hand extreme of this acceptance angle will appear to come from the loudspeaker on the left and similarly sound on the other extreme will appear to come from the right-hand side. This result will be obtained, no matter how far away the sounds are from the microphone. This fixed angle of acceptance, which depends on the type of microphone pair used, can have a number of disadvantages. For example, with a simple " single-microphone " balance of an orchestra there will be one position of the microphone where the orchestra occupies the full width of the reproduced sound stage. If the microphone is placed nearer than this some instruments will be in the out-of-phase areas of the microphone polar characteristic (Fig. 16.3), and if the microphone is further away the orchestra will only take up part of the available

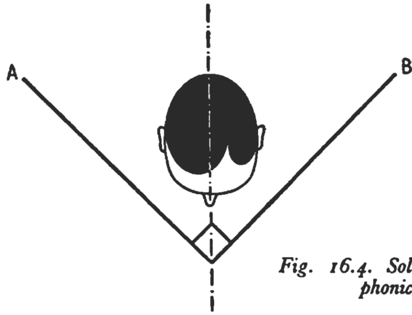


Fig. 16.4. Soloist very close to stereophonic microphone pair

sound stage. Thus there may be difficulty in reconciling a suitable width of reproduced sound with a satisfactory direct/indirect sound ratio. Furthermore, if a soloist is placed very close to a stereophonic microphone, the effect would be that the soloist would be reproduced having an exaggerated width (Fig. 16.4). This is obviously not required in normal circumstances, but may be of use for a special dramatic effect.

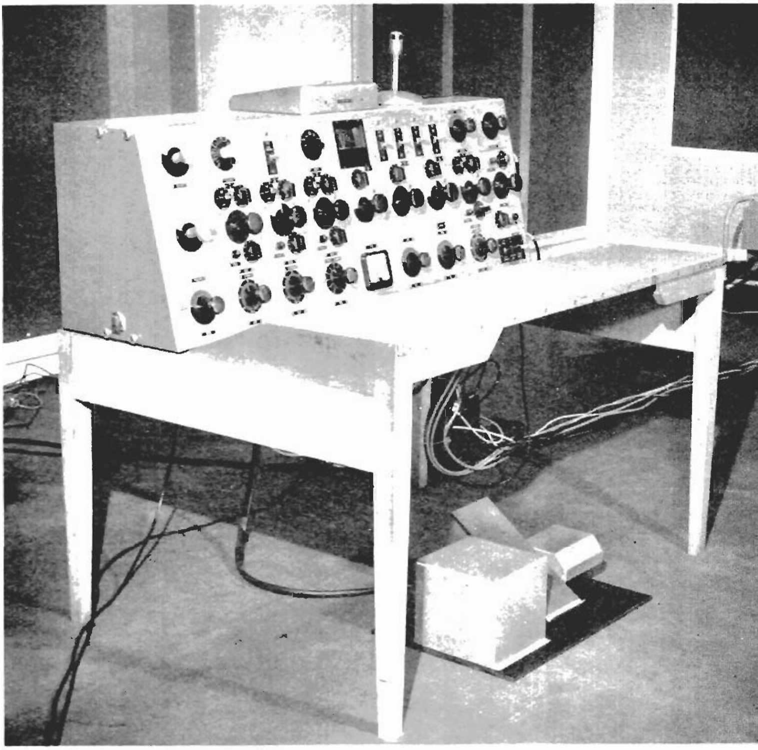
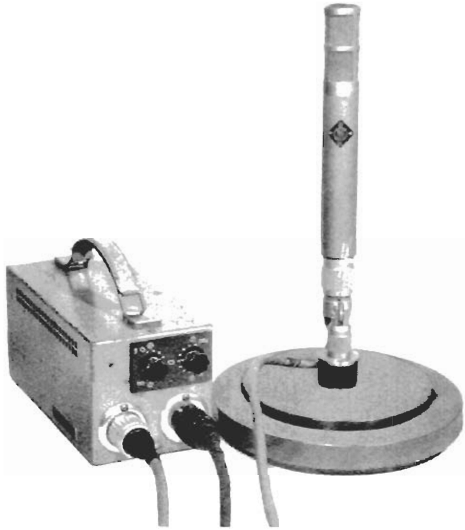


Plate 16.1. BBC experimental stereophonic studio control desk



Above: Plate 16.2. Neumann SM2 stereophonic microphone. Below: Plate 16.3. AKG C.24 stereophonic microphone



Fortunately it is possible, by electrical means, to modify the scale of width of the reproduced sound. To understand how this can be done, let us for a moment consider what would happen if we were to connect the two channels of one stereo microphone together. In this case, whatever information was in the left-hand channel will now also appear in the right-hand channel and vice-versa, so that both channels will have identical inputs. Our stereo " picture " will have collapsed to a point source half-way between the two

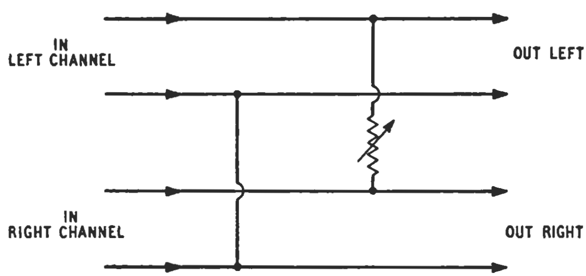


Fig. 16.5. Simple width control—narrowing only

loudspeakers. If now, instead of " shorting together " the two channels in this way, we connect a variable attenuator between them, it should be possible by altering the value of this attenuator to achieve any picture width from the full distance between the loudspeakers at maximum attenuation, to a point source in the centre at minimum (Fig. 16.5).

The foregoing statements are only true when both channels are " in phase ". Indeed, the supposition that both channels are in phase is necessary to all the arguments so far in this chapter. If the channels were not in phase, no centre image would be possible and an unpleasant effect would be produced. To return to our width control, if instead of connecting our attenuator so that it " shorts " the channels in phase, we were to reverse the connections to one channel, so that as the attenuator was varied the two channels were shorted out of phase, we would find that a certain amount of widening of the stereo picture is possible. It is obviously not possible to widen it to a very great extent, because when the channels are shorted out completely in the out-of-phase condition the stereo effect disappears. The limited amount of widening that is possible, however, can prove useful when the stereo microphone has to be placed at such a distance from the instrumentalists that

their image in the sound picture is rather small. Such an occasion might arise in a studio with a rather dead acoustic where, in order to achieve a satisfactory ratio of direct to indirect sound, the microphone distance might need to be rather large.

In practice, it is technically difficult to make a control which can be used for both widening and narrowing the sound picture when it is necessary to have a phase switch at some position in the travel of the control. Fortunately, another solution is possible. In the narrowing case, one might consider that shorting channels " in phase " is the same as adding them, and in the widening case, shorting the channels " out of phase " is the same as subtracting one from the other. Hence one might say that widening and narrowing can be achieved by varying the amount of sum or difference signal present in the two channels. It is a simple matter using transformers with double wound secondary windings to produce the sum

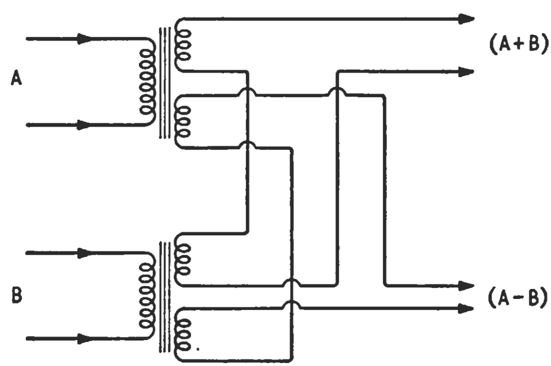


Fig. 16.6. Sum and difference circuit

signal and the difference signal from our two stereo channels (Fig. 16.6). A suitable control can then be devised which will vary these two signals—our width control. The sum and difference signals can then be combined with transformers as before in order to extract the two channel signals again. Putting this another way, if A and B are left- and right-hand signals:—

$A + B =$ sum (" M " signal in international terminology)

$A - B =$ difference (" S " signal in international terminology)

These are our two signals to vary with the width control (Fig. 16.7). On recombining and taking the average:—

$$\frac{(A+B)+(A-B)}{2}=A$$

$$\frac{(A+B)-(A-B)}{2}=B$$

A secondary effect occurs during the operation of the width control, which will also affect the ratio of direct to indirect sound.

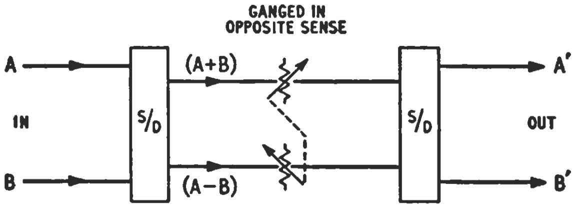


Fig. 16.7. Width control in sum and difference signals

As the picture width is narrowed, less reverberation will be reproduced, and as it is widened, reverberation will be increased. This is explained by the fact that it can be shown that the " S " signal contains most of the reverberant sound.*

16.3.1. Solo and Spot Microphones

From the above it will be evident that for a normal solo or spot microphone a stereophonic microphone is not suitable. The solution to the problem is to use a monophonic microphone for the soloist and to connect its output into both channels in such a way that the image from this microphone coincides with the weak image of the soloist on the main microphone. The device used for this operation is known as a " panoramic potentiometer " (Fig. 16.8), the name arising from the fact that by its use a sound source can be moved at will to any position in the width of the sound picture. In the use of a spot microphone, care must be taken to see that there is no confusion between the images due to this microphone and the main microphone. The setting-up procedure is as follows: the solo microphone channel gain control is advanced to such a point

* See section on " M-S " Stereophony at the end of this chapter

that the effect of the spot microphone can be heard, the " pan. pot." is moved until the image from this microphone coincides with that from the main microphone, and the gain control is adjusted to give the correct balance. A check on whether the two images have been correctly overlaid is made by fading quickly from the stereophonic microphone to the solo spot microphone. There should, of course, be no movement of the image. This principle can be

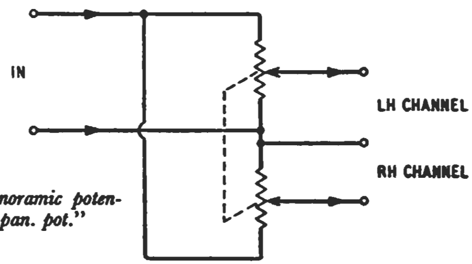


Fig. 16.8. Panoramic potentiometer—"pan. pot."

extended to an almost infinite number of monophonic or spot microphones and, in fact, in dance band and light music balances where multi-microphone technique has become a part of the sound produced, the use of a number of spot microphones is essential to produce the required effect. It should be stressed that at all times the stereophonic main microphone is essential in order to produce an overall reverberation or acoustic in which the sound can be set.

It would obviously be possible, indeed it has been done, to use a number of spot microphones, each producing a point image of an instrument or group of instruments, spaced across the sound picture by means of " pan. pots.", and dispense with the stereophonic microphone altogether. This, however, does not produce such a convincing result, since no sound appears in the space between the instruments, whereas, of course, in the studio, this space would be occupied by reverberation. A more convincing reproduction is achieved, therefore, if the stereo microphone is used as a basis and the spot microphones used for reinforcement only.

A further complication must be avoided. It is possible for any two spot microphones to act as a spaced pair of stereophonic microphones and thus produce spurious images of sounds within their pick-up area. Care must be taken in siting spot microphones to reduce this effect to a minimum, or blurring of the reproduced sound stage will result.

16.3.2. Use of More Than One Stereophonic Microphone

Whilst the foregoing argument applies strictly to solo instruments which are small in physical size, for example, solo violin and solo flute, there may be other occasions when the solo instrument is rather larger, like a piano, or indeed is not a soloist at all but a chorus, accompanied by an orchestra. This chorus might not be larger than the orchestra itself, but comparable in size, and placed to one side or the other. A monophonic microphone in front of such a chorus would produce only a point source of sound, and clearly this is not what is required. In this case a stereo chorus microphone is necessary in order to give width to the chorus and some means must be found of placing the image of the chorus due to its microphone in the same place as the image on the main microphone, and also of ensuring that the width of the chorus is again the same as the width from the main microphone. Fortunately this can be done. Some side movement of the stereo microphone output can be achieved by actual turning of the microphone. In addition, to a limited extent a movement similar to the " pan. pot." in the case of the monophonic microphone can be obtained by altering the relative gain of the left and right halves of the microphone. This can be termed " off-set ". We have already seen how the width control can be used to adjust the width of chorus in the complete montage.

16.4. ARTIFICIAL REVERBERATION FACILITIES

In any studio control equipment used for music or drama some means of adding artificial reverberation is necessary, and stereophonic equipment is no exception. In this case, as well as the usual mixture switch, some other facilities can be provided. Let us assume that the reverberation is provided by a room. The reverberation room loudspeaker need not be paired since one loudspeaker is all that is necessary to excite reverberation in the room. The microphone, however, must be a stereophonic one, since we may wish to create reverberation over the whole sound picture.

The feed for the loudspeaker can be taken from four different sources:—the left-hand channel, the right-hand channel, the sum, or the difference of the two. The return from the reverberation room microphone can also be made in several ways:—to the left-hand channel alone, the right-hand channel alone, stereo, the sum of the two to both, or even to a " pan. pot." This last will produce reverberation from any part of the reproduced sound stage. By combining these functions in different ways, many effects can be

produced. For instance, in light music, if the strings are on the left-hand side of the orchestra, the left-hand channel can be fed to the reverberation room, and the string reverberation thus produced returned to the studio control desk as stereo, when it will be heard over the whole of the sound picture. If it were returned on the left-hand channel only, the reverberation would appear as a point source at the extreme left of the picture. In a dramatic production, it is possible to make someone walk from the left-hand side to the right-hand side, being in a reverberant acoustic on the left and a dead one on the right. It is also possible, by feeding the reverberation room loudspeaker from the difference signal, to make him walk from, apparently, a reverberant corridor on the left-hand side of the picture through the centre of the stage in a dead acoustic and out through a similar corridor on the right.

16.5. ANCILLARY EQUIPMENT

On a studio desk designed for stereo operation, it will by now be evident that there will be a large number of individual controls (Plate 16.1). For each stereo channel there will be a fader, a width control, a reverberation mixture switch, a reverberation source selector, an off-set control for moving the whole of the stereo picture to right or left, and one other control. This last is a pre-set control used during the setting up process for each studo session to ensure that the two halves of each microphone chain are perfectly balanced, including of course the stereophonic microphone itself, the most likely source of unbalance. Each spot microphone channel will have its fader, a " pan. pot." and a reverberation mixture switch. Facilities will also be provided for the playing in of stereo effects, probably by tape, and for connecting monophonic disc recorded effects to one or more spot channels.

Monitoring is, of course, necessary, and a P.P.M. having two pointers, one for each channel, has been developed. Loudspeaker feeds must, of course, be provided with suitable arrangements for controlling their volume simultaneously. Since it is likely that the " M " or sum signal of the two channels will be used as the signal for the monophonic listener to a stereophonic programme, a switch must be provided to feed this signal to one loudspeaker in order that it may be checked.

This " M " signal should contain all the information necessary for the monophonic listener, but care must be taken that the sound produced is as close as possible to that from a normal monophonic

balance. This necessary condition may be difficult with some programme material.

16.6. STEREOPHONIC RECORDING

Probably the most convenient method of recording twin-channel stereophony is by means of magnetic tape. In this case, the two tracks can be laid side by side, each track occupying half of the tape, as is favoured in professional practice. Alternatively, as in some domestic recorders, each track may occupy one quarter of the tape, four tracks being used—two in each direction to save tape—effectively extending the " half track " system to stereo (Fig. 16.9). Few problems exist in the tape recording of stereo, though it is of course essential that both recording and reproducing amplifiers in the stereo machine be as identical as possible. In the early days of stereo recording, the two half-track heads were placed one after the other on the recording machine with the result that one track was displaced by a distance depending on the head spacing, in relation



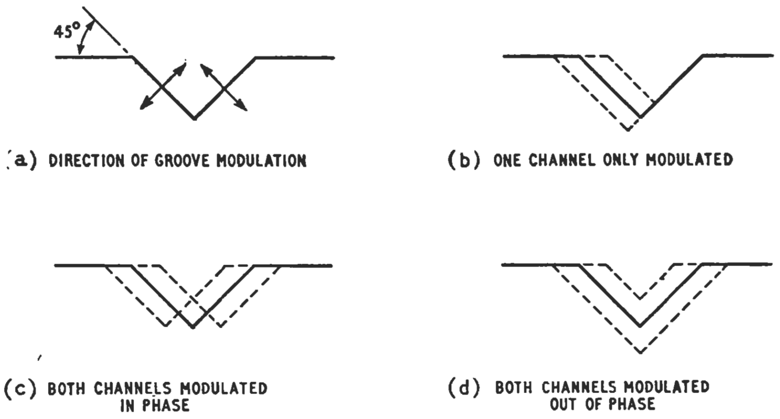
Fig. 16.9. (a) ½ track stereophonic tape recording. (b) ¼ track stereophonic tape recording tracks 1 and 3 in one direction of tape, 2 and 4 in the other

to the other. This had the disadvantage that the head spacing could well be different with different makes of machine, and the more modern practice of stacking the two heads one above the other is much more satisfactory, and has now become standard. Standardisation has also been introduced as to which track is associated with which channel. With the tape playing from left to right with the magnetic coating away from the observer, the top track should be associated with the left channel. This applies with both half- and quarter-track trecording.

Whilst magnetic tape recording is undoubtedly the most satisfactory method for master recording and broadcasting work,

it is only slowly being accepted in the home and stereophonic disc recording on a fine-groove long-playing disc is becoming increasingly popular as a means of domestic reproduction of pre-recorded stereo sound. In this case the two tracks are both recorded in the same groove on the disc, effectively one in each wall of the groove, each track being cut at 45° to the plane of the disc. A complex cutter has been designed to perform this function, and the cutter head is fed in such a manner that with two signals in phase, a lateral cut is produced

Fig. 16.10. Section through groove of stereophonic disc recording



on the record and with the signals out of phase, a vertical (see Fig. 16.10). Reproducing pick-ups likewise are designed to respond to the two sets of modulation and to produce two distinct outputs, one for each channel. Great care must be exercised in the design of both recording and reproducing heads to ensure that the cross-talk is kept at a minimum. Because of the complex nature of the recorded groove the reproducing stylus tip needs to have a rather smaller radius than that commonly used for monophonic long-playing records. This smaller tip radius necessitates a lighter playing weight for the pick-up if undue harm is not to be done to the surface of the record, since the pressure on the disc increases as the radius decreases, according to a square law.

16.7. RADIO TRANSMISSION OF STEREOPHONY

The radio transmission of stereophony has presented more difficult problems than has the recording of the stereo signal. An important requirement is that if a stereo programme is transmitted

by radio, a monophonic listener tuned to the same transmission must be able to hear the programme in a normal manner, i.e. his reception should not be impaired by the fact that the programme is also being broadcast stereophonically. This " compatibility ", whilst obviously essential in a radio service, proves to be difficult where transmission systems are concerned since, in achieving it, other undesirable factors become evident.

The simplest method of transmitting stereophony by radio, and undoubtedly the most efficient in terms of pure stereophonic reproduction, so far as only one service area is concerned, is by using one complete transmission chain for each channel. Obviously this is highly inefficient in terms of economics, and in the use of a limited number of transmission channels. Moreover, under these conditions neither transmitter is radiating an acceptable monophonic signal, i.e. the system is not compatible. The experiments in stereophonic broadcasting so far carried out by the BBC have been using this system, but work is going on to develop a compatible system which will enable the two stereophonic channels to be carried by one radio transmitter, using only one V.H.F. channel.

Various systems for achieving this have been described, but so far all of them have disadvantages of one form or another, and have not been really suitable. Further development is necessary. A brief description of some of these systems now follows.

16.7.1. E.M.I. " Percival " System

The system proposed by E.M.I. is an ingenious one which would enable stereophonic transmission to be carried not only by one transmitter, but also by one land line, an important point when the type of programme distribution used in the United Kingdom is considered. In this case, no matching of two land lines, one per channel, would be necessary, thus saving considerable expense and technical difficulty. The system works briefly as follows: the two channels are added together and their sum is transmitted by the radio system in the normal way as a monophonic signal. Experiment has shown that this sum signal produces, in most cases, an acceptable signal for the monophonic listener. In addition to the sum signal, a further signal derived from the two channels, termed the " directional " signal, is also transmitted along the land line and by the transmitter. This is possible because the directional information supplied by the system can be compressed into a bandwidth of less than 100 c/s, and this can conveniently be transmitted at the upper end of the audible spectrum without impairing the

programme material. At the receiving end, the sum signal is applied to both channels and the directional information is extracted and applied to the two channel amplifiers via electrical networks in such a way that it controls the relative gain of the two amplifiers. Thus it will be seen that with the left-hand amplifier at zero gain and the right-hand amplifier at maximum, the signal will appear to come from the right of the picture, and similarly with the right-hand amplifier at zero gain and that on the left at full gain the sound will appear on the left. At gain settings between these limits sounds can be made to appear at any point in the sound picture. The supposition is that there is a " persistence of hearing " akin to the well-known phenomenon of " persistence of vision ", and that providing that the directional information acts quickly enough, a complete sound picture will be built up. In effect, the directional information acts like a high speed " pan. pot.", moving the sound almost instantaneously to its correct location. Unfortunately, as was described earlier in the section on microphone technique, the end result is in some ways similar to the use of a number of " pan. pots." with no overall stereo microphone, i.e. there is a lack of an enveloping acoustic. Various other limitations are evident in the system at present, but research and development are still proceeding.

16.7.2. The Crosby System

The system, proposed by the Crosby Corporation in the U.S.A., is intended primarily for use with a V.H.F. radio transmitter and again the sum of the two channels is transmitted in the normal manner of a monophonic programme. This system can therefore be considered to be compatible. In this case, however, no directional signal in the Percival sense is derived, but the difference signal (see argument on sum and difference above) is transmitted on a sub-carrier along with the main transmission. This sub-carrier can be of the order of 50 kc/s and the difference signal is frequency modulated upon this. At the receiving end electrical networks separate the sum and difference signals, pass them through a further sum-and-difference or " matrixing " network, and produce the left and right signals again. This system is capable of producing good stereophonic reproduction, and indeed has been used for stereophonic experiments in the United States and in Holland. Unfortunately, however, because the total bandwidth of the transmission is restricted the radiated sum signal is at a lower level than it

would be if it were radiated without the sub-carrier signal. Consequently the signal-to-noise ratio at the receiver is worse, by some decibels, than in the monophonic case. This will, of course, affect the monophonic listener. The stereophonic listener will also be affected in this respect since the sub-carrier channel has an even worse signal-to-noise ratio than the main channel. The matrixing network averages this noise out over both channels, but the overall effect can be anything up to 20 dB worse than the normal monophonic transmission. This system can work very well within the primary service area of the transmitter, as is the case with most American local broadcasting stations, but the signal-to-noise problem becomes acute when the high-powered broadcasting system with large fringe areas, such as that used in the United Kingdom, is attempted. Another difficulty with this system is that the co-channel and adjacent channel interference may be worse due to the presence of the sub-carrier.

16.7.3. The Mullard System

The system proposed by the Mullard Company also enables the two channels to be carried on one V.H.F. transmitter. In this case the transmitter modulation input equipment is switched at a very high speed, some 32,000 times per second, from one channel to the other so that left and right channels are transmitted alternately by the one transmitter. The monophonic listener has no means of separating these and so hears effectively the sum of the two channels, a compatible signal. The stereophonic listener has an electronic switch synchronised with the switch at the transmitter, and is thus able to separate the two channels and feed them to their respective loudspeakers. Again the stereophonic effect is well transmitted but signal-to-noise and interference problems are still present.

16.7.4. United States System

The Federal Communications Commission (F.C.C.) of the United States has now approved a system for stereophonic broadcasting by stations in that country. This system is practically identical to those proposed by the Zenith and General Electric Companies of the U.S.A. and is a multiplex system for radiation by VHF, FM transmitters.

The main channel carries the " compatible " sum or " M " signal, and a sub-carrier, the difference or " S " signal. Unlike the Crosby system described above the sub-carrier is *amplitude*

modulated, and the sub-carrier itself is suppressed, only the side-bands being radiated. A second pilot sub-carrier, unmodulated, is radiated at low level to ensure locking of the receiver circuits. The "stereophonic" sub-carrier is at 38 kc/s, and the pilot sub-carrier at 19 kc/s.

In addition to the stereophonic sub-carriers, the system allows for the inclusion of one or more other frequency modulated sub-carriers, designed for "storecasting".

Whilst such a system has been found suitable for broadcasting from small local stations, as in the U.S.A., it may be less suitable for the system of nation-wide coverage used in the United Kingdom and some other European countries. The problems of signal-to-noise ratio, and interference between neighbouring channels still exist.

16.8. MICROPHONES FOR STEREOPHONY

Whilst two ribbon microphones of the PGS or 4038 type can be used for co-incident microphone stereophony, it is difficult to place them close enough together because of their physical size, and to maintain a balance of frequency response between them over the required spectrum. Furthermore, the figure-of-eight characteristic does not always produce the best possible effect, and so the polar response should preferably be adjustable. Special microphones have been developed for use in stereophony, mainly of the electrostatic type where the relative frequency response of the two capsules can be controlled more accurately; and their small size enables two to be mounted very close together. Two microphone capsules and two microphone amplifiers can be conveniently mounted in one case very little larger than the equivalent monophonic microphone.

16.8.1. **Neumann SM.2** (Plate 16.2)

This is a small stereophonic microphone using two capsules, each similar to that used in the KM56 monophonic microphone. Provision is made for rotating one capsule in relation to the other to enable them to be set at the required angle. Variable polar diagram control is provided for both capsules by means of separate controls on the power unit.

16.8.2. **AKG C24 Microphone** (Plate 16.3)

As might be guessed, this is in fact a combination in one case of two C.12 microphones as described in Chapter 5. Again provision

is made for rotating one microphone capsule in relation to the other and the complete range of polar characteristics available with the C.12 is present with each capsule.

16.9. M/S SYSTEM

An alternative system of microphone technique has also been used, which will give the " M " and " S" signals directly, without the necessity of sum and difference networks. In this case the two microphones have different polar characteristics, as shown in Fig.

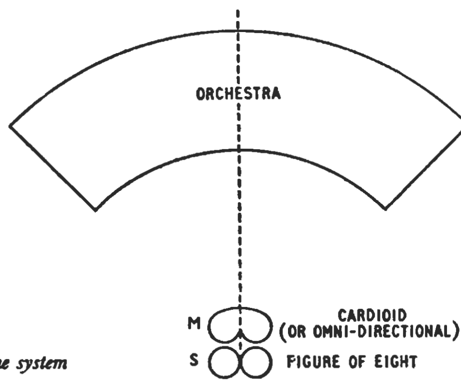


Fig. 16.11. M/S microphone system

16.11. One is a forward facing cardioid or an omni-directional microphone, and the other a figure-of-eight with its live axis at 90° to the general direction of the sound source.

The forward-facing microphone provides the " M " signal, and the sideways figure-of-eight the " S " signal. It will be seen immediately that little direct sound should reach the " S " microphone, almost the entire output of this microphone being due to the reverberant sound in the studio.*

It has been suggested that since this arrangement generates the " M " and " S " signals directly a saving of equipment in a transmission system can result. Unfortunately it is still necessary to extract the *A* and *B* signals for monitoring purposes so this saving seems to be an illusion.

The effectiveness of the stereophonic or derived monophonic reproduction by this system is no different from that of the *A* and *B* system described earlier.

* Hence it becomes apparent that the increase or decrease of the " S " signal will alter the reverberation as well as the width of the picture (section 16.3).

APPENDIX

A.1. THE DECIBEL

In Chapter 2 a description is given of the use of the *decibel* to compare sound intensities. This unit is also used to denote electrical levels in the programme chain, the basis of comparison being the standard level of 1 milliwatt (0·001 watt) in a 600 ohm circuit. This is known as *zero level*, and other levels above or below this are quoted as $+x$ dB or $-x$ dB, the statement "above (or below) zero level" being taken for granted. Programme at about this level gives comfortable listening on headphones or telephones—hence its use in ring main circuits, and as the studio sending level.

A level of 0·1 watts, for example, would be quoted as + 20 dB,

$$\text{since } 10 \log_{10} \frac{0{\cdot}1}{0{\cdot}001} = 10 \log_{10} 100 = 20$$

The *gain* of amplifiers, and the *loss* of attenuators are also expressed in dB, the actual number of dB being calculated from the formula

$$\text{number of dB} = 10 \log_{10} \frac{P_1}{P_2}$$

where P_1 and P_2 are the input and output powers.

In most studio equipment, the *voltage* level is more important than the *power* level. To express the voltage gains etc. in dB a modified form of the formula is used.

If $\frac{P_1}{P_2}$ is the power ratio,

$P_1 = \frac{V_1^2}{Z}$ where V_1 is the voltage and Z the impedance in the circuit.

Also, $P_2 = \frac{V_2^2}{Z}$

$$\therefore \frac{P_1}{P_2} = \frac{V_1^2}{V_2^2} = \left(\frac{V_1}{V_2}\right)^2$$

Hence number of (voltage) dB $= 20 \log_{10} \frac{V_1}{V_2}$

This shows that the number of voltage dB is twice the power dB for a given ratio, and the reason is that the power ratio in a given circuit is proportional to the square of the voltage ratio. (This is only strictly true if the impedances at the two points are equal.)

Examples

(i) How many dB separate 500 watts and 5 watts?

Number of dB $= 10 \log_{10} \frac{500}{5} = 10 \log_{10} 100 = 20$ dB.

(ii) What is the gain of an amplifier which gives an output of 8 volts for an input of 0·0008 volts?

$$\text{Number of (voltage) dB} = 20 \log_{10} \frac{8}{0{\cdot}0008}$$

$$= 20 \log_{10} 10{,}000 = 80 \text{ dB}$$

(iii) How many watts of electrical power correspond to + 40 dB with reference to zero level?

$$40 = 10 \log_{10} \frac{P}{0{\cdot}001}$$

$$\therefore \log_{10} 1{,}000\, P = 4$$

$$\therefore 1{,}000\, P = 10^4$$

$$\therefore P = \frac{10{,}000}{1{,}000}$$

$$= 10 \text{ watts}$$

Table A.1.

POWER AND VOLTAGE dB

Ratio	*Power dB*	*Voltage dB*
2 : 1	3	6
4 : 1	6	12
10 : 1	10	20
20 : 1	13	26
100 : 1	20	40
1,000 : 1	30	60
10,000 : 1	40	80
100,000 : 1	50	100
1,000,000 : 1	60	120

A.2. SIMPLE HARMONIC MOTION

If we let a point P move round a circle at a uniform speed, the angle " α " will take up all values from 0° to 360°, and y will rise to a maximum in one direction, fall to zero, reach maximum in the

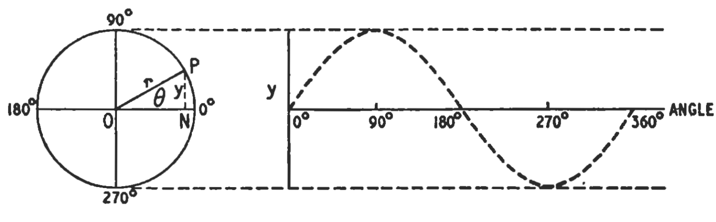


Fig. A.1. The sine-wave graph of Sin θ against θ

opposite direction, and again fall to zero. Drawing a graph, plotting y against the angle θ, we obtain a smooth curve (Fig. A.1).

Two features should be noted:—

(a) At all positions of P, OP = r, the radius of the circle, and if we let $r = 1$, we see that plotting θ against y is equivalent to plotting θ against sin θ. This curve is therefore called a *sine wave.*

(b) As P moves round the circle with a uniform velocity, M moves to and fro, imitating the bob of a pendulum or the prong of a tuning-fork. This motion is called *Simple Harmonic Motion*, and is a characteristic of a pure tone. Thus all the particles of a tuning-fork take up such a vibration. This is communicated to all the air particles in the vicinity, and to the eardrums of an observer.

The distance of a vibrating particle from its mean position x at any instant is called its *displacement*.

The maximum displacement is called the *amplitude*—symbol a. A complete vibration of N through all positions and back again is called 1 *cycle*—and is related to 1 circle of revolution of P.

The stage which a particle has reached in its cycle is called its *phase*. Thus crests are in phase, and a crest and a trough are out of phase by 180°.

Note: The horizontal scale in Fig. A.1 might equally well be a *time* scale.

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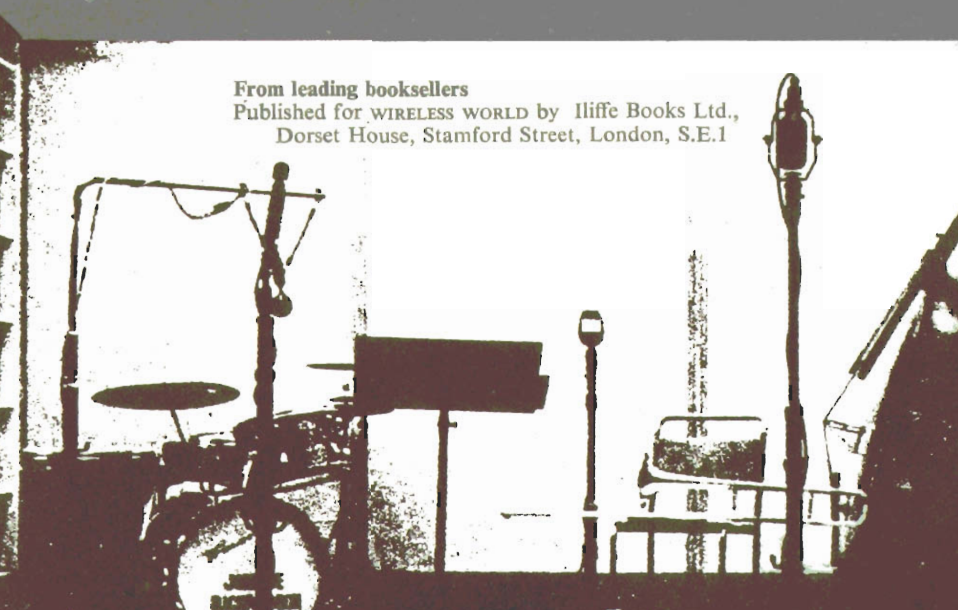
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